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Product Guide

IP Telephony Product Lineup

The Next Wave of Telecommunication Innovation!

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Product Manager / Product Marketing Dept.

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Are Your Communications Implemented with VoIP?

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"How much do you pay for long distance and international phone charge?"

"How many staffs do you need to maintain data and telephone network?"

"How many types of wire are in your company?"

"How much TCO (Total Cost of Ownership) do you pay for these?"

Build Your IP Communication Platform with Micronet IP Telephony Solution!

Protocol: SIP (Session Initiation Protocol)

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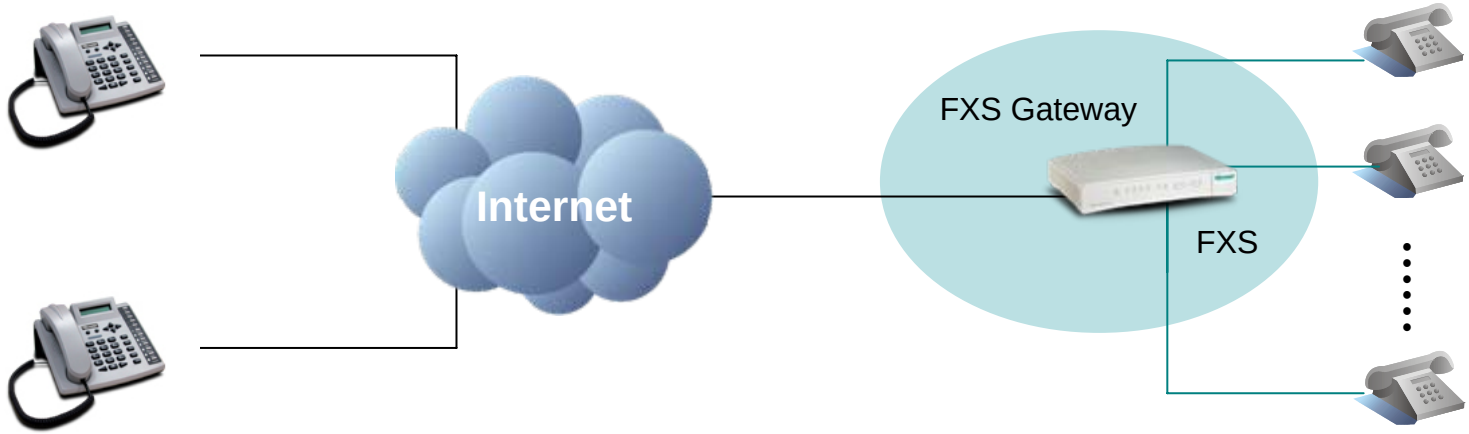
- Specified by IETF SIP Working Group. It was to provide a signaling and call setup (initiate, modify, terminate) protocol for IP-based communications
- SIP is the next great protocol of Internet because it brings an entirely new set of media and collaboration capabilities to applications, "converged applications", including instant messaging, voice/video telephony, video conferencing, rich-media enterprise collaboration, click-to-call…
- In developing SIP-based applications and networks, it got simplicity, interoperability, scalability, fault tolerance… rather than H.323 that has architecture designed for "Old World", PSTN (circuit-switched network).

Voice Interface: FXS

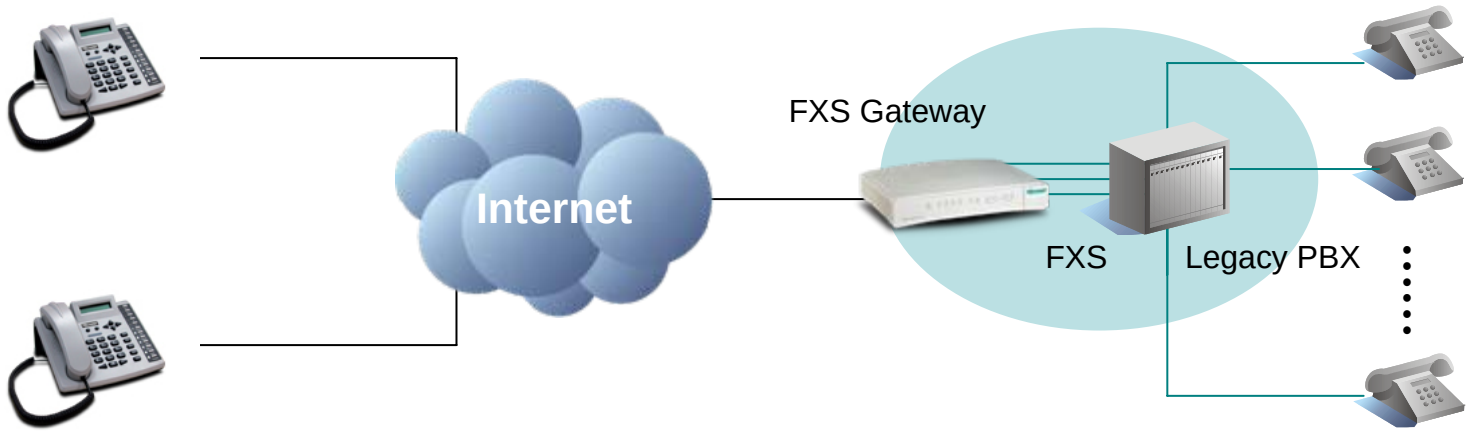
- An FXS interface, in essence, mimics the PSTN switch (call exchange center)
- An FXS interface can be used to connect basic devices, such as PBX’s trunk, phones and faxes
- Provide voltage, off-hook detection, call progress indicators (dial tone, disconnect tone, hold tone…), and ring generation

Voice Interface: FXS

- FXS port connecting standard telephone for direct VoIP dial out



- FXS port connect PBX's trunk for VoIP upgrade in larger scale

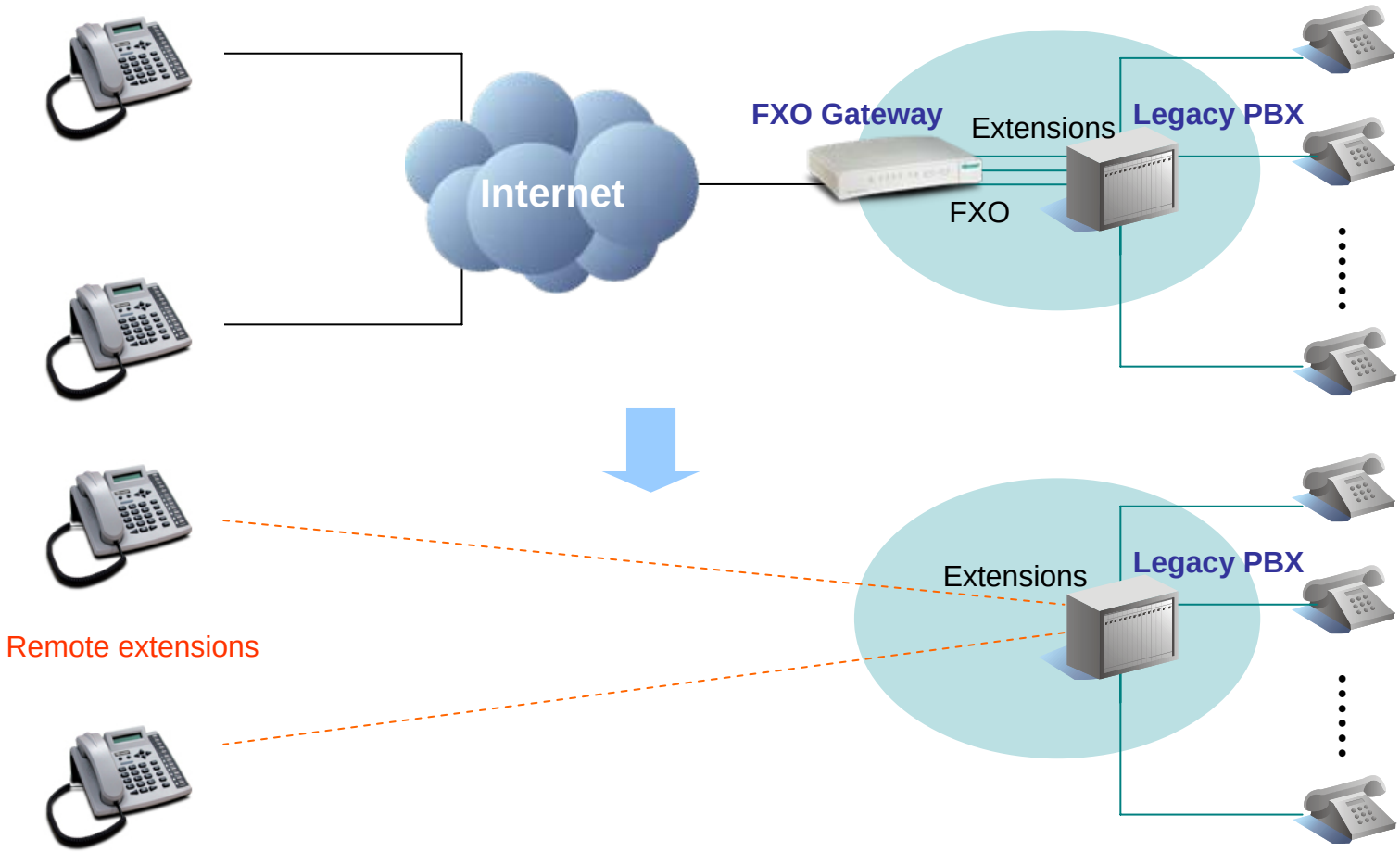


Voice Interface: FXO

- An FXO interface is like the port on a standard telephone
- An FXO interface can be used to connect PBX's extension or PSTN
- Receive/Transport calls from and to a PBX or PSTN (Public Switched Telephone Network)
- Requires a dial tone, ring indication, and call progress indicators that are provided from PBX or PSTN switch
- Automate the operations (on-hook, off-hook) that are done manually on a standard telephone

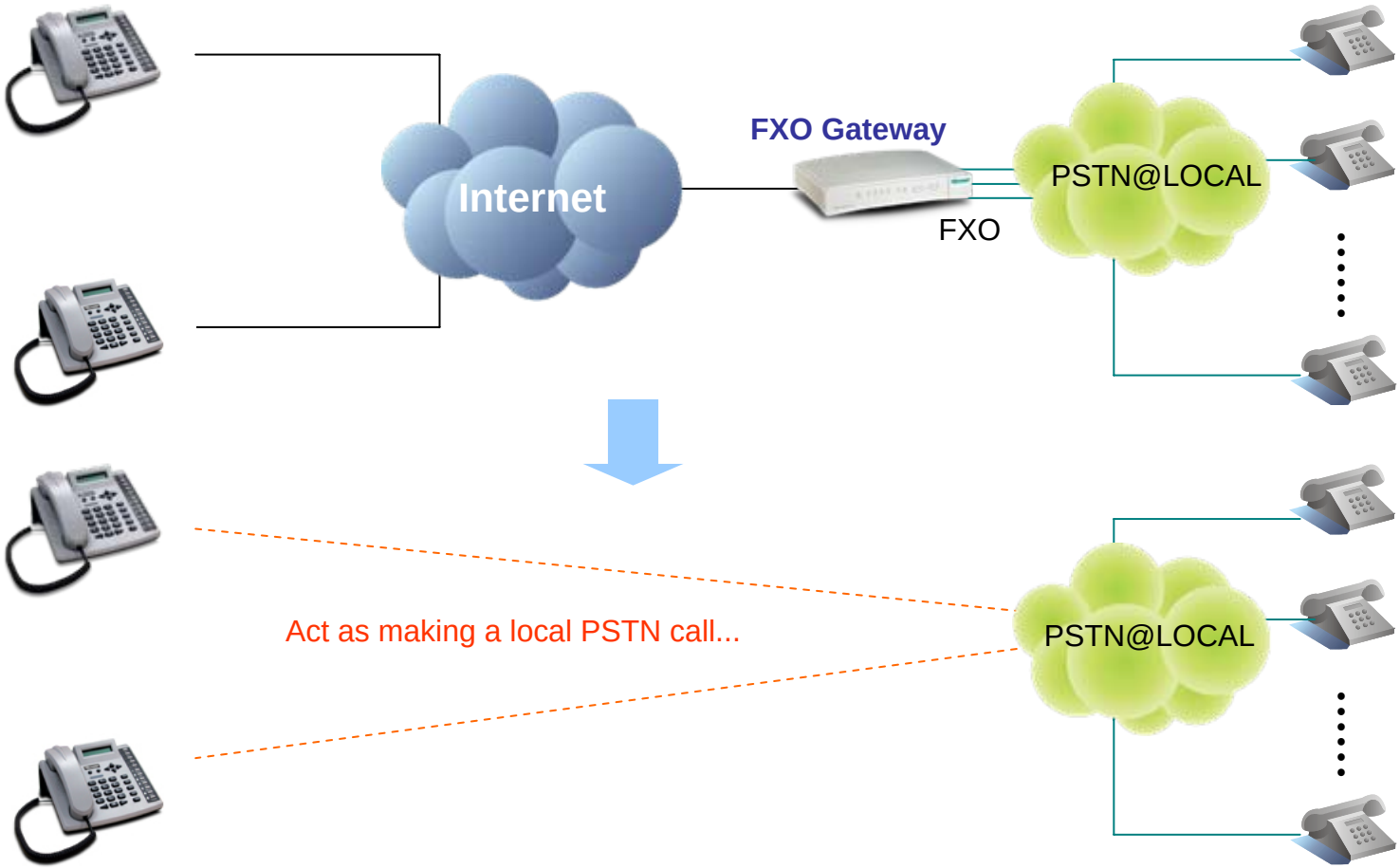
Voice Interface: FXO

- Connecting FXO ports to the extension ports of PBX will set IP Phone (or other VoIP device) as remote extensions



Voice Interface: FXO

- Connecting FXO ports to PSTN will allowing IP Phones to dial from international sites with local PSTN charge only



Voice Interface: E1/T1 PRI

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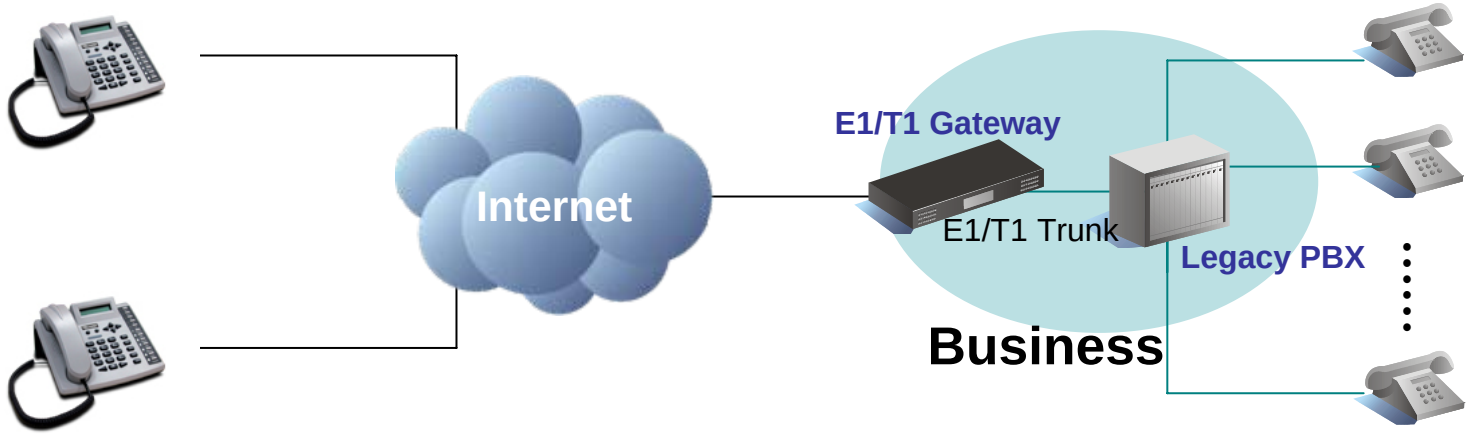
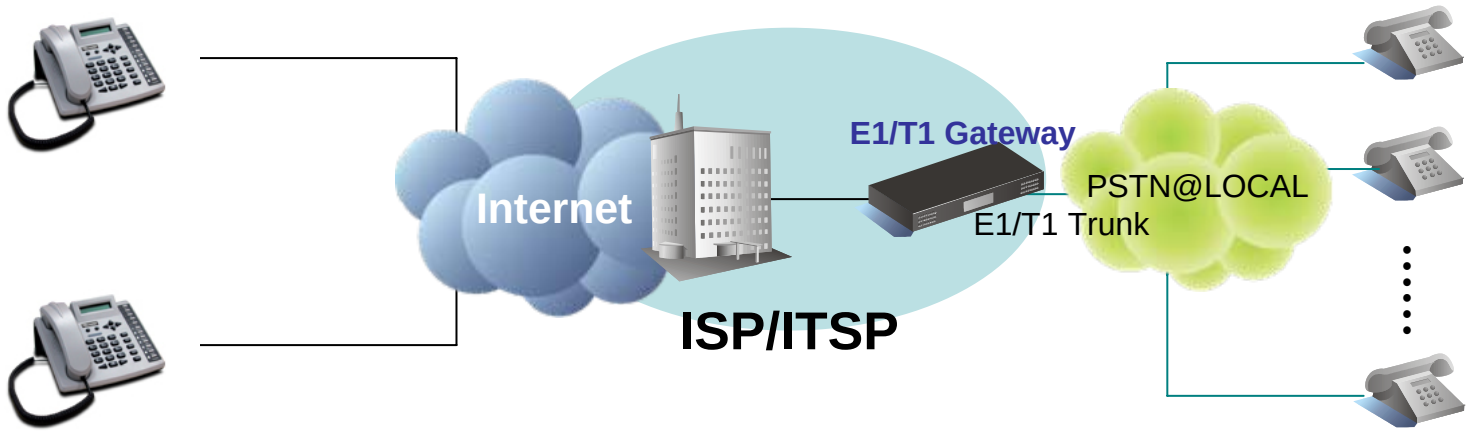
- One E1/T1 (digital) line consists of 30/23 Bearer channels (voice) and 1 Data channel (signaling)
- Signaling is controlled by PSTN protocols, such as E1 ISDN PRI, E1 SS7 ISUP, E1 CAS DTMF/ R2, T1 ISDN PRI, T1 CAS
- Used to connect with PSTN or legacy PBX
- To meet the demand for huge-volume call traffic

Voice Interface: E1/T1 PRI

- E1/T1 port connecting legacy PBX for big enterprise (business) implementation

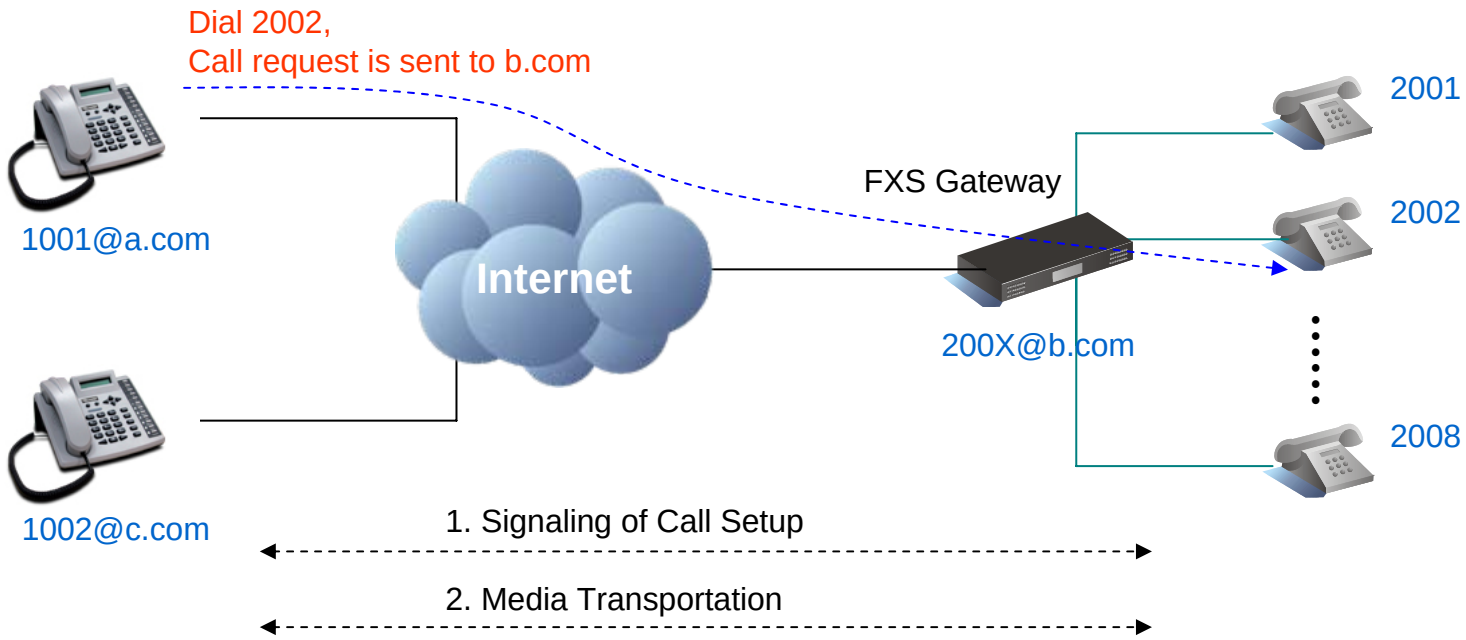


- E1/T1 port connect PSTN for ITSP implementation



VoIP Operation Mode: P2P

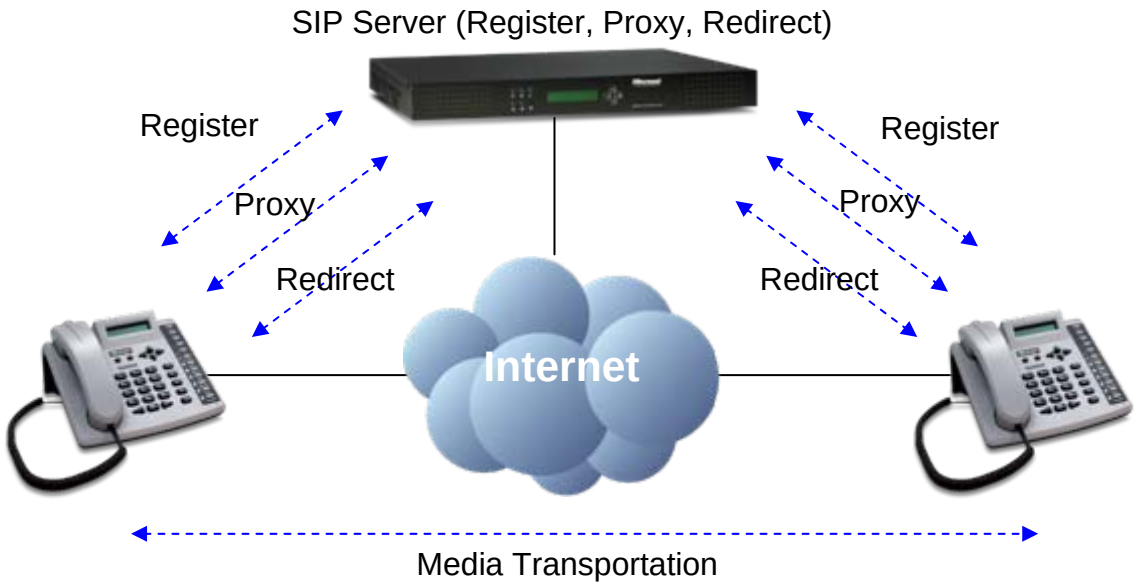
- Call setup is completed between 2 VoIP clients, without independent server
- Signaling are transmitted through the path that is configured in the entry of "Dial Plan" and "Phone Address Book".



- Public and fixed IP address (or URL) is necessary to each VoIP device.
- Not good when the number of VoIP devices is big.

VoIP Operation Mode: Proxy

- Register: Support VoIP clients to register and keep the records of TEL number and associated IP address
- Proxy: Server routes the call request to called user
- Redirect: The alternative to Proxy. Server sends back the address of called user and helps caller to locate called user



VoIP Operation Mode: Proxy

- Server-grade device, such as SIP Proxy Server and IP PBX, is needed to exchange signals of call setup between caller and called users.
- Sometimes for NAT pass-thru of media (RTP), media has to be routed through the server, when the client is located behind router and using private IP address.
- Server supports clients with telephony features, such as call forward, call transfer, call hold, digit insert/delete …
- VoIP CPE (endpoint) can work anywhere it can connect to Internet and register to server

IP PBX Basics

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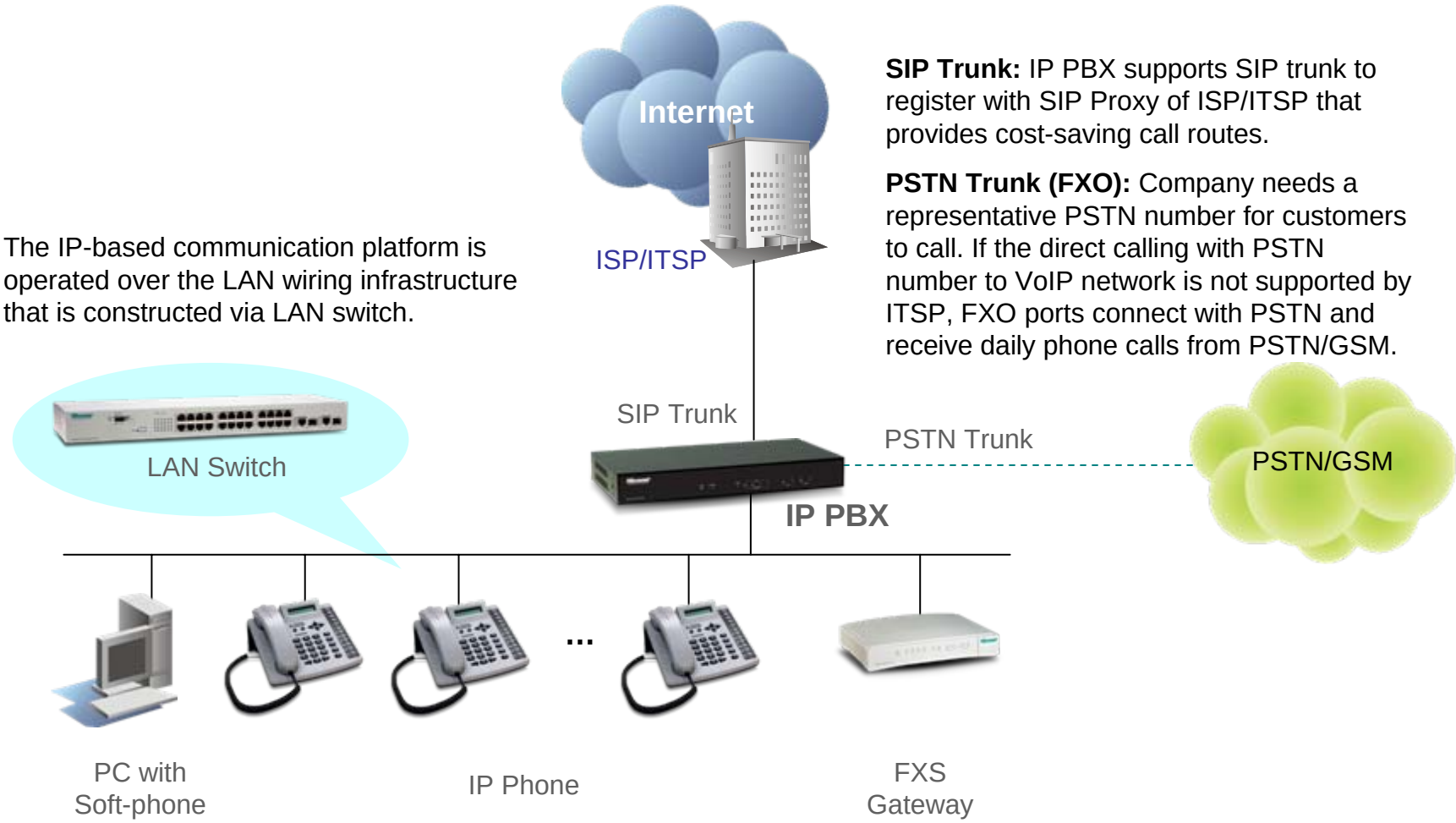
- Server-grade Device: IP PBX provides VoIP clients (eg. IP Phone, FXS gateway) with registration, SIP call proxy/redirection, call control, call routing, etc.
- Cost-Saving: IP PBX supports SIP trunk to ITSP for cost-saving call routes.
- Full Functions as Legacy PBX: It also includes automated attendant, IVR (Interactive Voice Response), voice mail system, and other value-added functionalities to fulfill user's needs that have been served by legacy PBX.
- Pure IP Implementation: IP PBX is used for Greenfield to build a pure IP-based communication platform.
- Upgrade from Legacy: With built-in FXS ports, IP PBX is also good for legacy PBX system to upgrade in the aspects of scale and functionality (VoIP).

IP PBX Basics

The IP-based communication platform is operated over the LAN wiring infrastructure that is constructed via LAN switch.



SIP Trunk: IP PBX supports SIP trunk to register with SIP Proxy of ISP/ITSP that provides cost-saving call routes.

PSTN Trunk (FXO): Company needs a representative PSTN number for customers to call. If the direct calling with PSTN number to VoIP network is not supported by ITSP, FXO ports connect with PSTN and receive daily phone calls from PSTN/GSM.

Codec vs. Bandwidth

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Codec	Bit Rate	MOS	Bandwidth Consumption
G.711	64 kbps	4.1	87.2 kbps
G.723.1	5.3 kbps	3.9	20.8 kbps
G.723.1	6.3 kbps	3.8	21.9 kbps
G.726	32 kbps	3.85	55.2 kbps
G.728	16 kbps	3.61	31.5 kbps
G.729	8 kbps	3.92	31.2 kbps

- The mean opinion score (MOS) provides a numerical measure of the quality of human speech at the destination end of the circuit.

Micronet IP Telephony Solution

CPE (Customer-Premise Equipment) Device	Carrier-Grade Device
IP PBX - Pure IP PBX, SMB - Pure IP PBX, ROBO - IP PBX Gateway **IP Phone** - Business IP Phone - Consumer IP Phone **Analog Gateway** - SIP FXS Gateway (ATA), 1/2 FXS - VoIP FXS Gateway, 2/4/8 FXS - ADSL2+ VoIP Router (IAD), 1 FXS - VoIP FXS/FXO Gateway, 2/4 FXS + 2/4 FXO - VoIP FXO Gateway, 2/4/8 FXO	**Digital Gateway** - E1/T1 Trunk Gateway, 1/4 Trunk **SIP Proxy Server** - SIP Proxy, 500 subscribers - SIP Proxy, 2,000 subscribers - SIP Proxy, 2,0000 subscribers

IP Telephony Case Study

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CATEGORY	CASE	CONTENTS	CHANNEL
SOHO VoIP	CASE1	IP Phone and ATA for Residential and SOHO Users	ISP/ITSP, Retail
	CASE2	IAD for Residential and SOHO Users	ISP/ITSP
SMB VoIP	CASE3	IP PBX Gateway for Trading Company, Travel Agency, or Small Business (under 10 users)	ISP/ITSP, SI
	CASE4	IP PBX Gateway for Trading Company, Travel Agency, or Small Business (under 10 users), with PSTN Backup	ISP/ITSP, SI
	CASE5	IP PBX for SMB Office (15 - 30 users)	SI, ISP/ITSP
	CASE6	IP PBX for SMB Office (50 - 70 users) with Call Routes via ISP/ITSP only	SI, ISP/ITSP
	CASE7	IP PBX for SMB Office (50 - 70 users)	SI, ISP/ITSP
	CASE8	Integrating FXS Gateway with legacy PBX for SMB Office (30 - 50 users)	ISP/ITSP, SI
	CASE9	Integrating IP PBX with legacy PBX for SMB Office (50 - 100 users)	SI, ISP/ITSP
Main/Branch Office VoIP	CASE10	VoIP Integration for Main/Branch Office	SI
	CASE11	VoIP Integration for Main/Branch Office, Minor Upgrade	SI
	CASE12	VoIP Integration for Main/Branch Office, Major Upgrade	SI, ISP/ITSP

Business Rule of Thumb

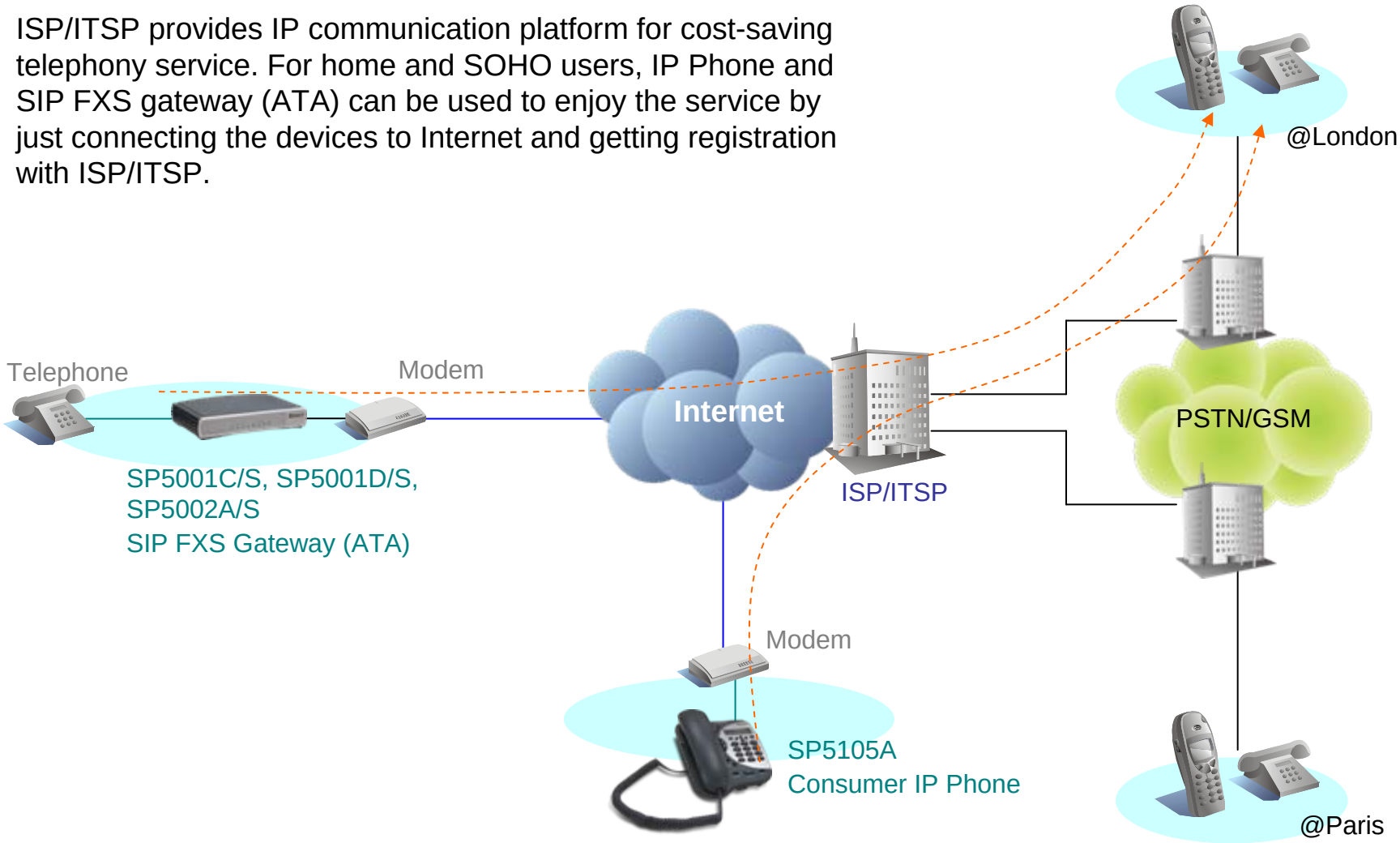
Micronet provides comprehensive IP Telephony product offerings.

They can be flexibly configured into many application suites,

While it is highly recommended to the VoIP System Integrators
to select some specific configurations carefully and sell them in focus.

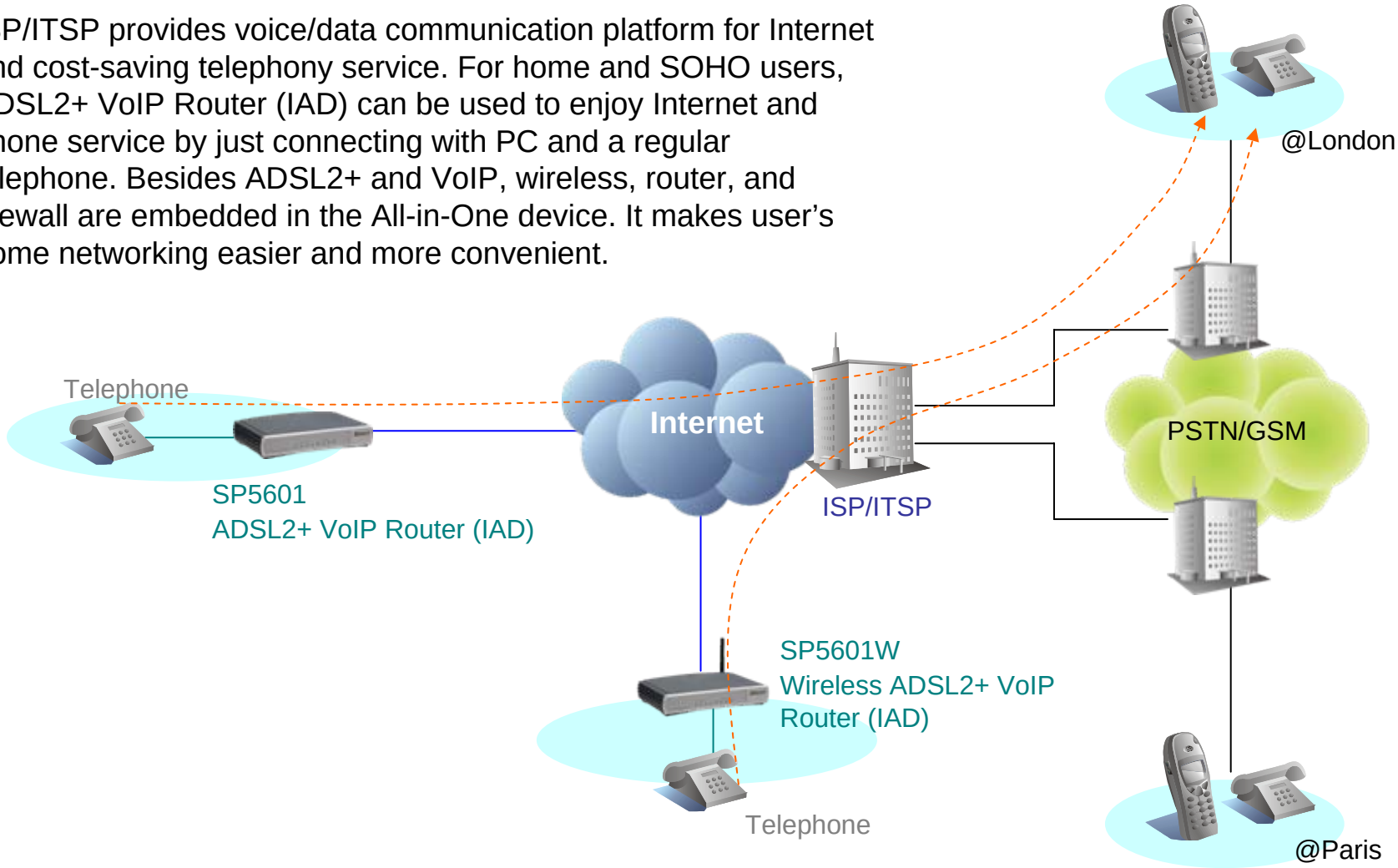
CASE 1: IP Phone and ATA for Residential and SOHO Users

ISP/ITSP provides IP communication platform for cost-saving telephony service. For home and SOHO users, IP Phone and SIP FXS gateway (ATA) can be used to enjoy the service by just connecting the devices to Internet and getting registration with ISP/ITSP.



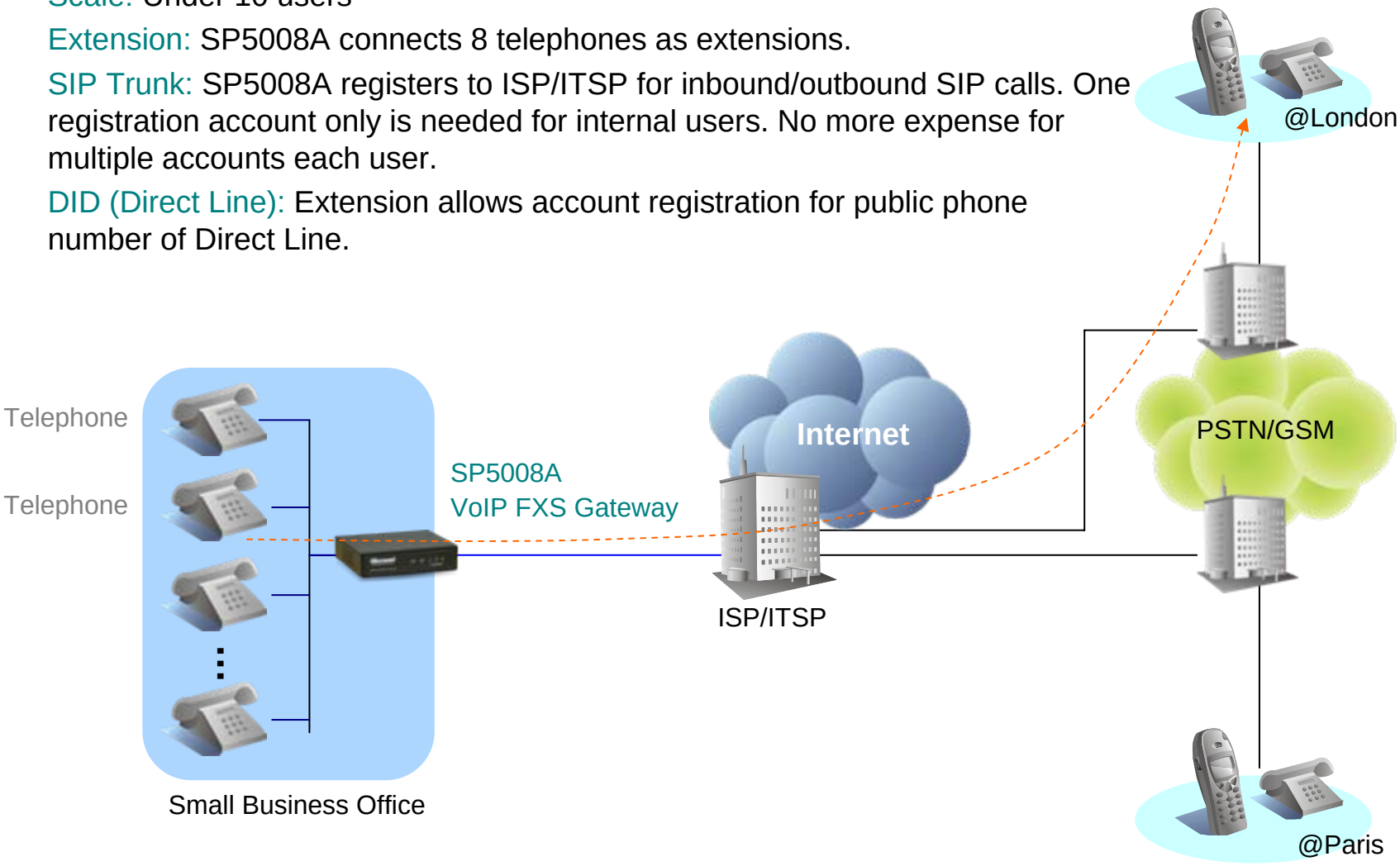
CASE 2: IAD for Residential and SOHO Users

ISP/ITSP provides voice/data communication platform for Internet and cost-saving telephony service. For home and SOHO users, ADSL2+ VoIP Router (IAD) can be used to enjoy Internet and phone service by just connecting with PC and a regular telephone. Besides ADSL2+ and VoIP, wireless, router, and firewall are embedded in the All-in-One device. It makes user's home networking easier and more convenient.



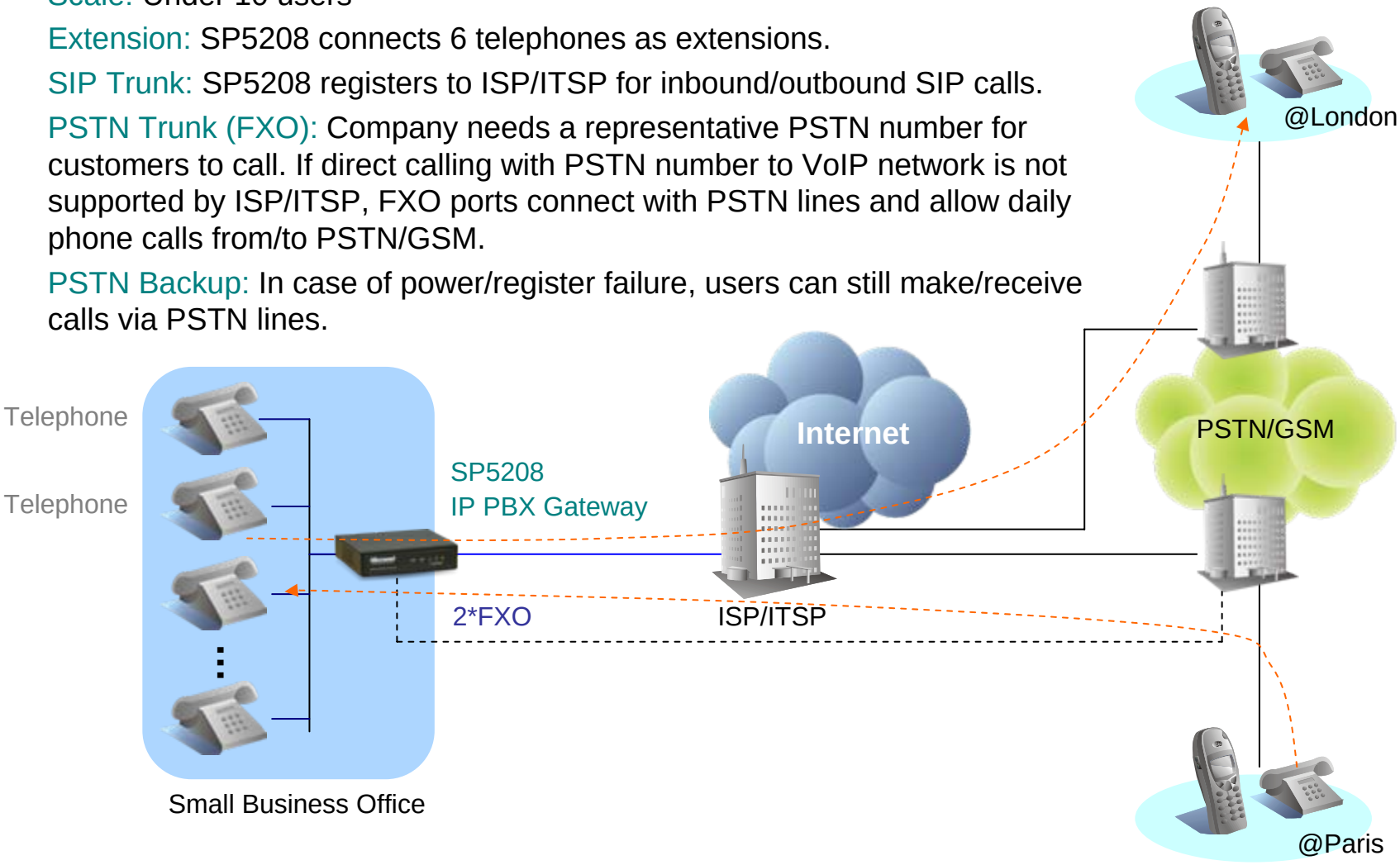
CASE 3: IP PBX Gateway for Trading Company, Travel Agency, or Small Business

- Scale: Under 10 users
- Extension: SP5008A connects 8 telephones as extensions.
- SIP Trunk: SP5008A registers to ISP/ITSP for inbound/outbound SIP calls. One registration account only is needed for internal users. No more expense for multiple accounts each user.
- DID (Direct Line): Extension allows account registration for public phone number of Direct Line.



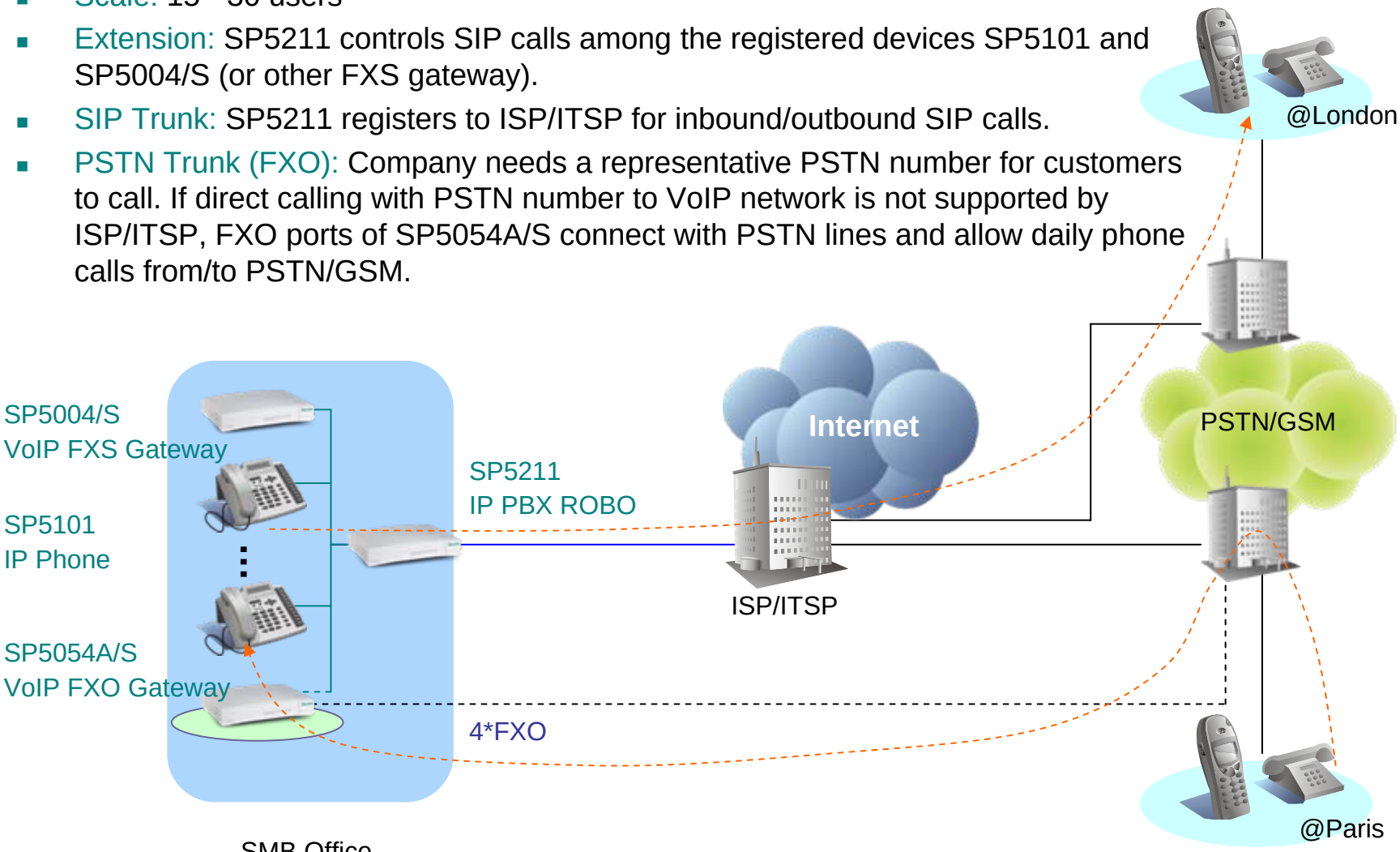
CASE 4: IP PBX Gateway (with PSTN Backup) for Trading Company, Travel Agency, or Small Business

- Scale: Under 10 users
- Extension: SP5208 connects 6 telephones as extensions.
- SIP Trunk: SP5208 registers to ISP/ITSP for inbound/outbound SIP calls.
- PSTN Trunk (FXO): Company needs a representative PSTN number for customers to call. If direct calling with PSTN number to VoIP network is not supported by ISP/ITSP, FXO ports connect with PSTN lines and allow daily phone calls from/to PSTN/GSM.
- PSTN Backup: In case of power/register failure, users can still make/receive calls via PSTN lines.



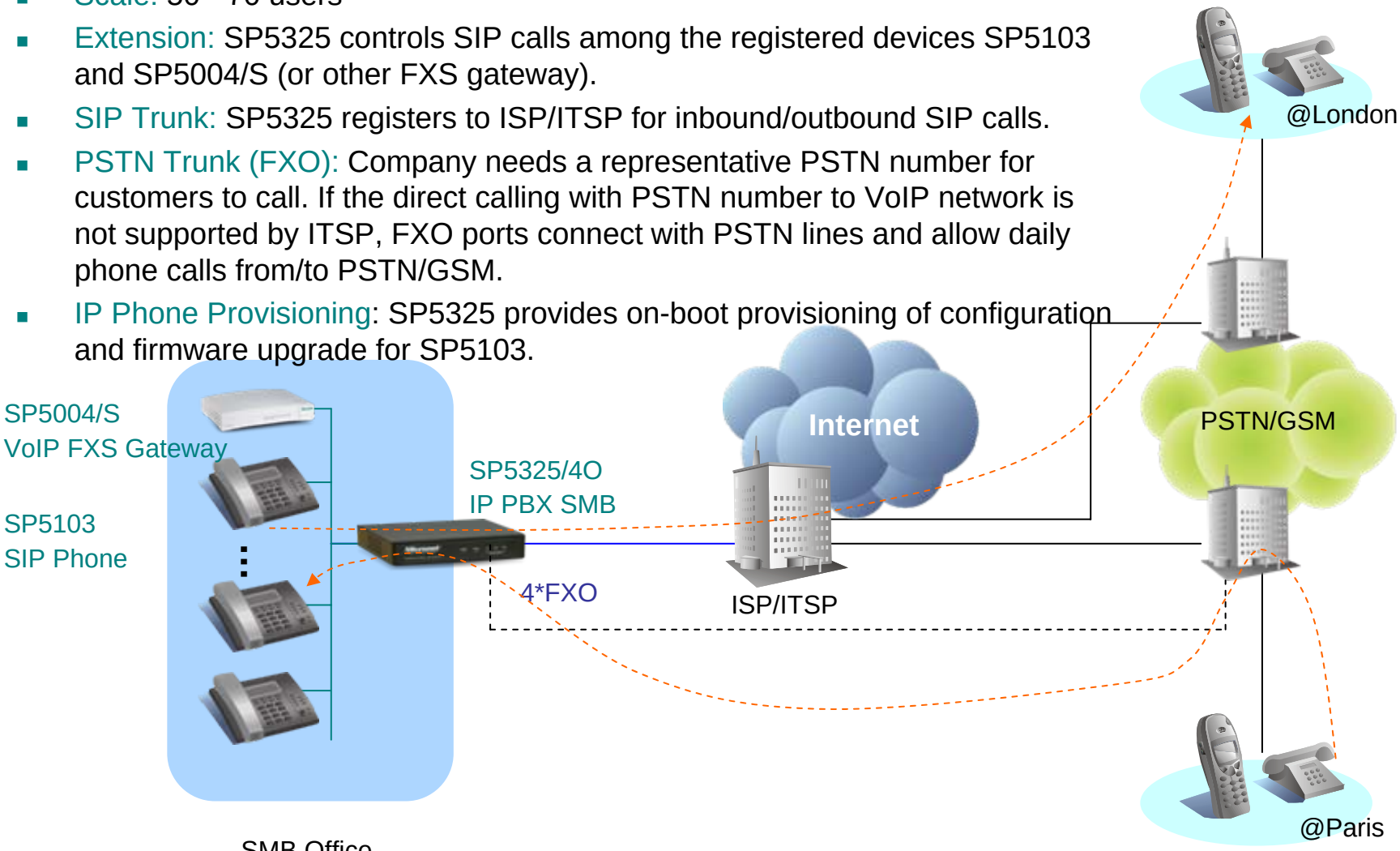
CASE 5: IP PBX for SMB Office

- Scale: 15 - 30 users
- Extension: SP5211 controls SIP calls among the registered devices SP5101 and SP5004/S (or other FXS gateway).
- SIP Trunk: SP5211 registers to ISP/ITSP for inbound/outbound SIP calls.
- PSTN Trunk (FXO): Company needs a representative PSTN number for customers to call. If direct calling with PSTN number to VoIP network is not supported by ISP/ITSP, FXO ports of SP5054A/S connect with PSTN lines and allow daily phone calls from/to PSTN/GSM.



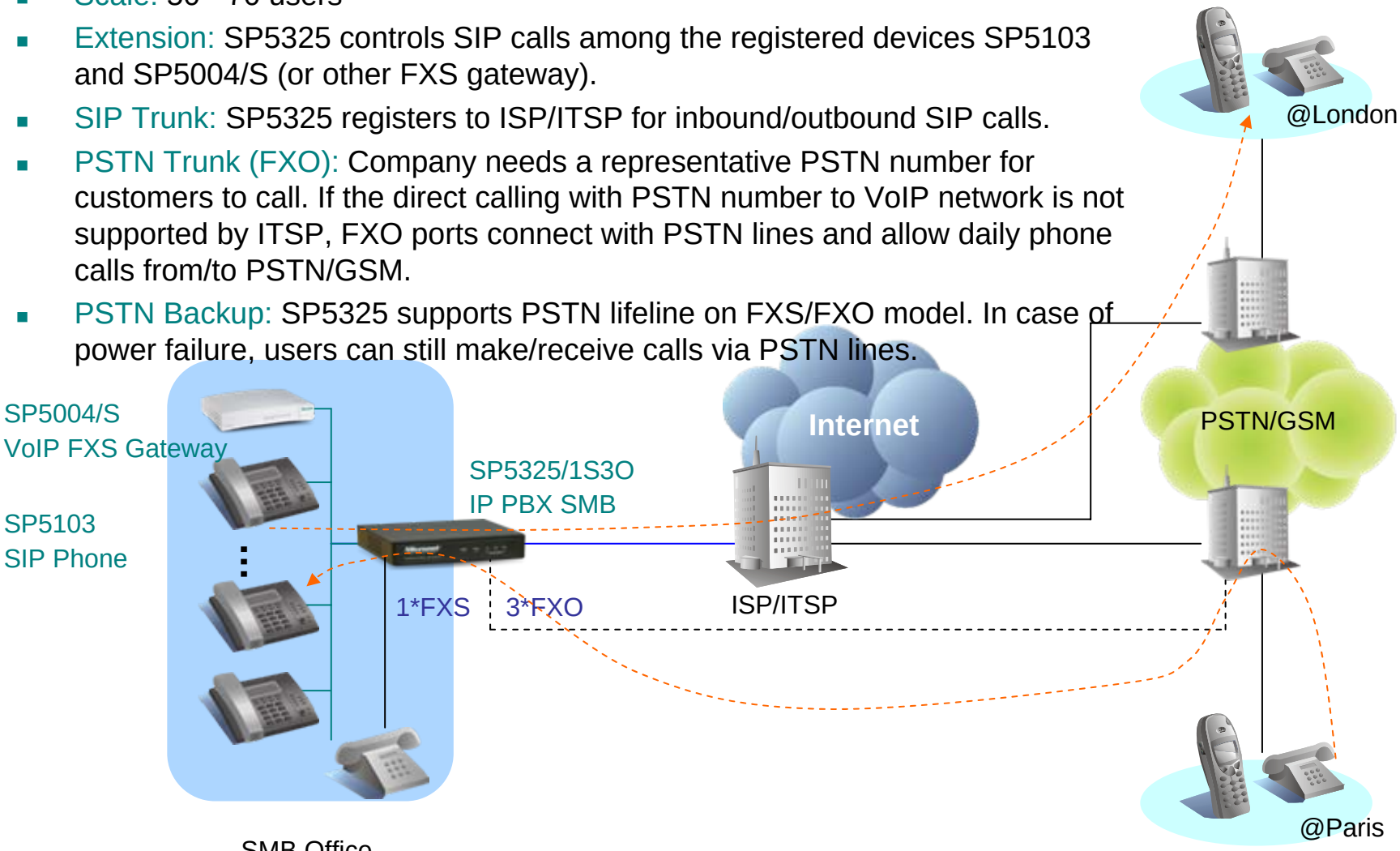
CASE 6: IP PBX for SMB Office

- Scale: 50 - 70 users
- Extension: SP5325 controls SIP calls among the registered devices SP5103 and SP5004/S (or other FXS gateway).
- SIP Trunk: SP5325 registers to ISP/ITSP for inbound/outbound SIP calls.
- PSTN Trunk (FXO): Company needs a representative PSTN number for customers to call. If the direct calling with PSTN number to VoIP network is not supported by ITSP, FXO ports connect with PSTN lines and allow daily phone calls from/to PSTN/GSM.
- IP Phone Provisioning: SP5325 provides on-boot provisioning of configuration and firmware upgrade for SP5103.



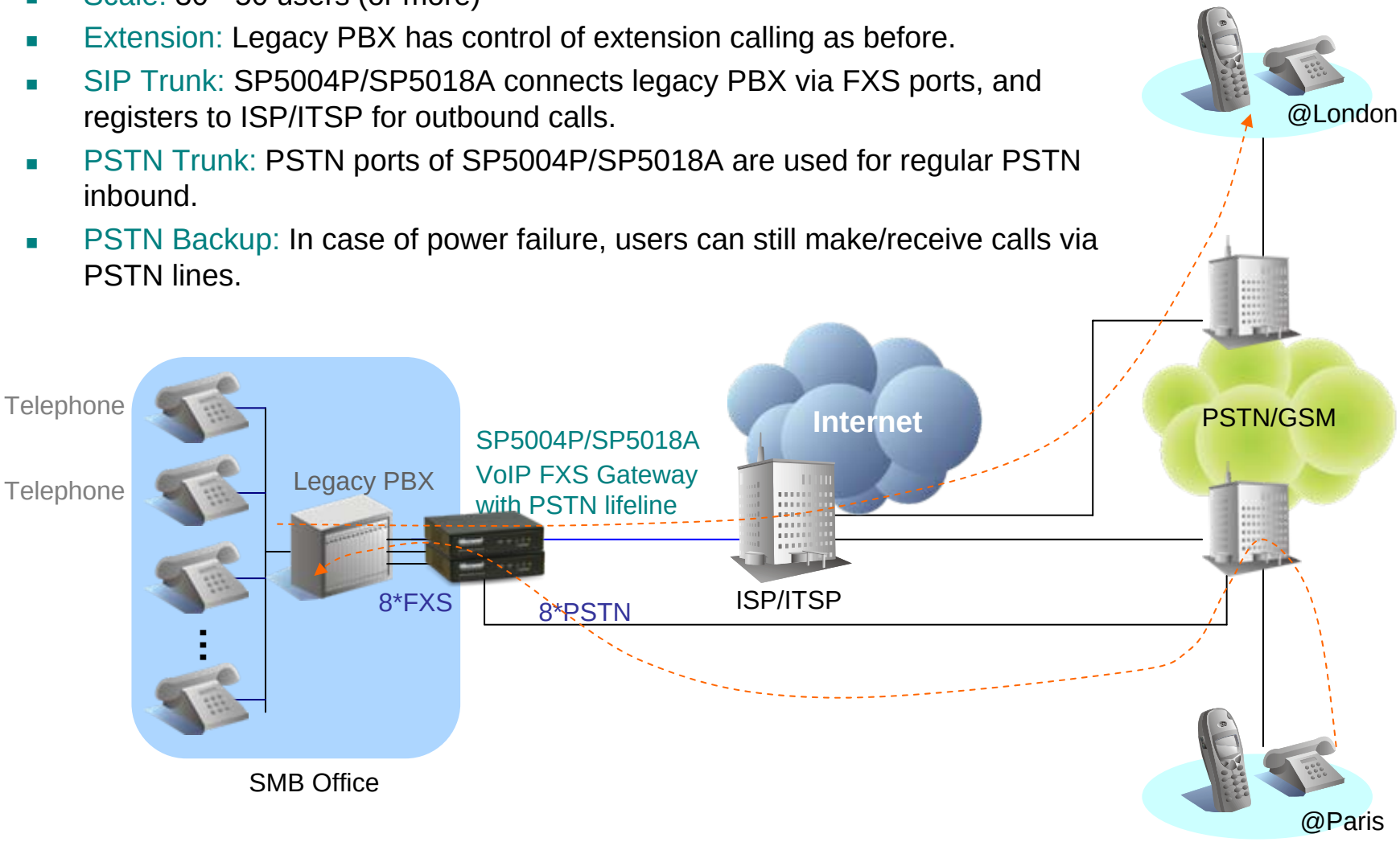
CASE 7: IP PBX for SMB Office

- Scale: 50 - 70 users
- Extension: SP5325 controls SIP calls among the registered devices SP5103 and SP5004/S (or other FXS gateway).
- SIP Trunk: SP5325 registers to ISP/ITSP for inbound/outbound SIP calls.
- PSTN Trunk (FXO): Company needs a representative PSTN number for customers to call. If the direct calling with PSTN number to VoIP network is not supported by ITSP, FXO ports connect with PSTN lines and allow daily phone calls from/to PSTN/GSM.
- PSTN Backup: SP5325 supports PSTN lifeline on FXS/FXO model. In case of power failure, users can still make/receive calls via PSTN lines.



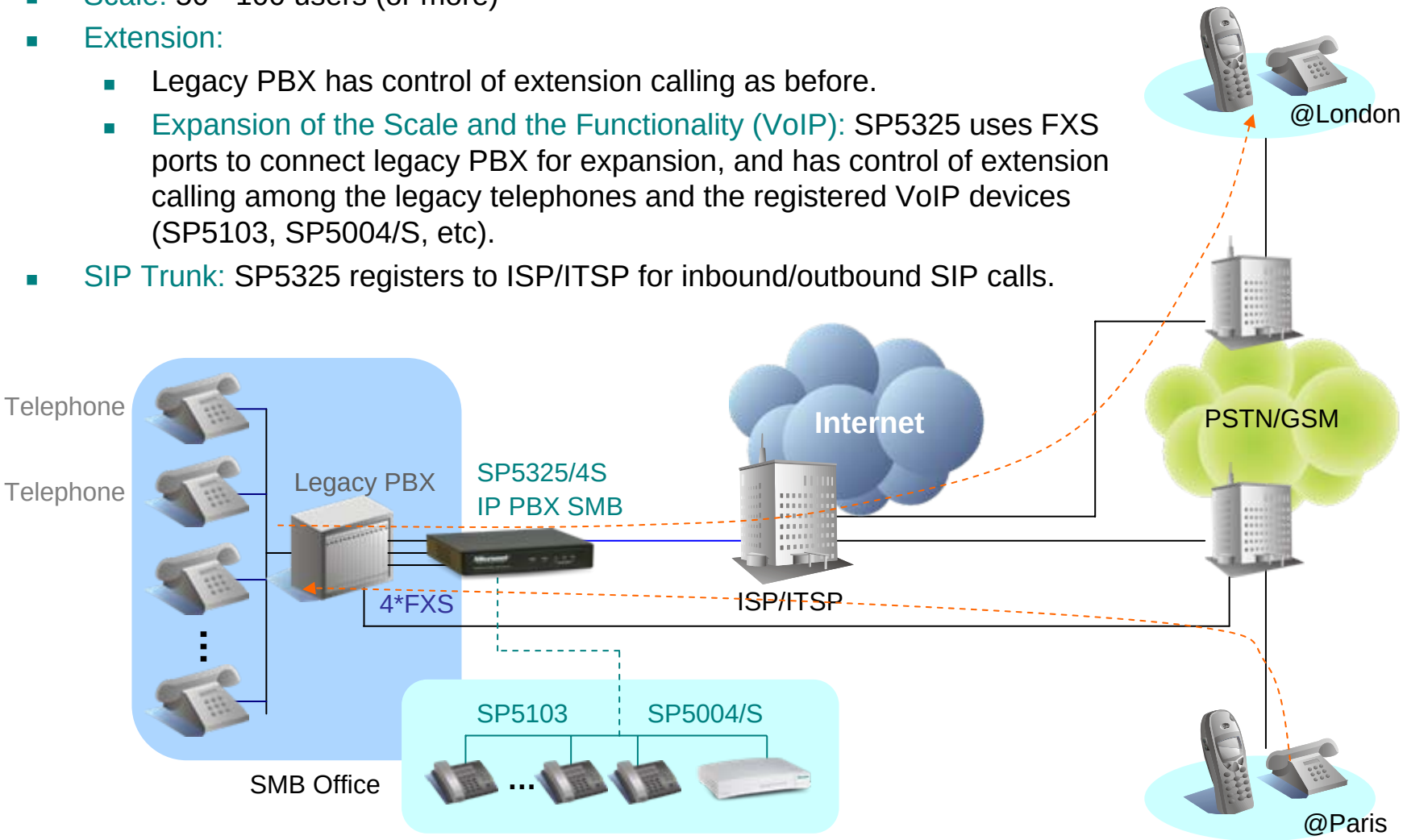
CASE 8: Integrating FXS Gateway with Legacy PBX for SMB Office

- Scale: 30 - 50 users (or more)
- Extension: Legacy PBX has control of extension calling as before.
- SIP Trunk: SP5004P/SP5018A connects legacy PBX via FXS ports, and registers to ISP/ITSP for outbound calls.
- PSTN Trunk: PSTN ports of SP5004P/SP5018A are used for regular PSTN inbound.
- PSTN Backup: In case of power failure, users can still make/receive calls via PSTN lines.



CASE 9: Integrating IP PBX with Legacy PBX for SMB Office

- Scale: 50 - 100 users (or more)
- Extension:
  - Legacy PBX has control of extension calling as before.
  - Expansion of the Scale and the Functionality (VoIP): SP5325 uses FXS ports to connect legacy PBX for expansion, and has control of extension calling among the legacy telephones and the registered VoIP devices (SP5103, SP5004/S, etc).
- SIP Trunk: SP5325 registers to ISP/ITSP for inbound/outbound SIP calls.

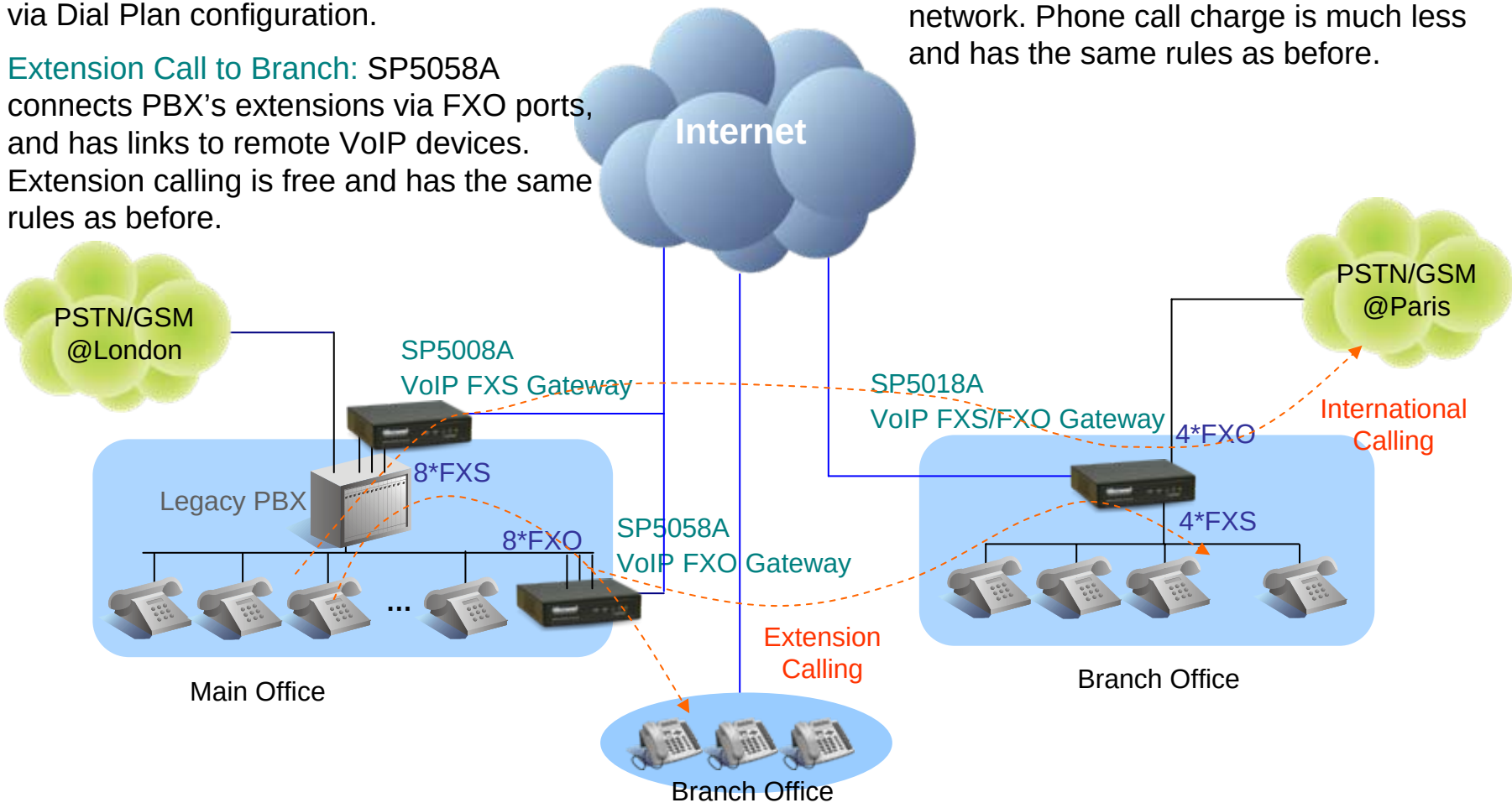


CASE 10: VoIP Integration for Main/Branch Office

Implementation: SP5008A, SP5058A, and SP5018A are deployed on multiple sites to integrate with POTS devices and have P2P connections to each others. Phone call is controlled via Dial Plan configuration.

Extension Call to Branch: SP5058A connects PBX's extensions via FXO ports, and has links to remote VoIP devices. Extension calling is free and has the same rules as before.

International Call: SP5008A connects PBX's trunk via FXS ports and has links to remote SP5018A to access remote PSTN/GSM network. Phone call charge is much less and has the same rules as before.



CASE 11: VoIP Integration for Main/Branch Office, Minor Upgrade

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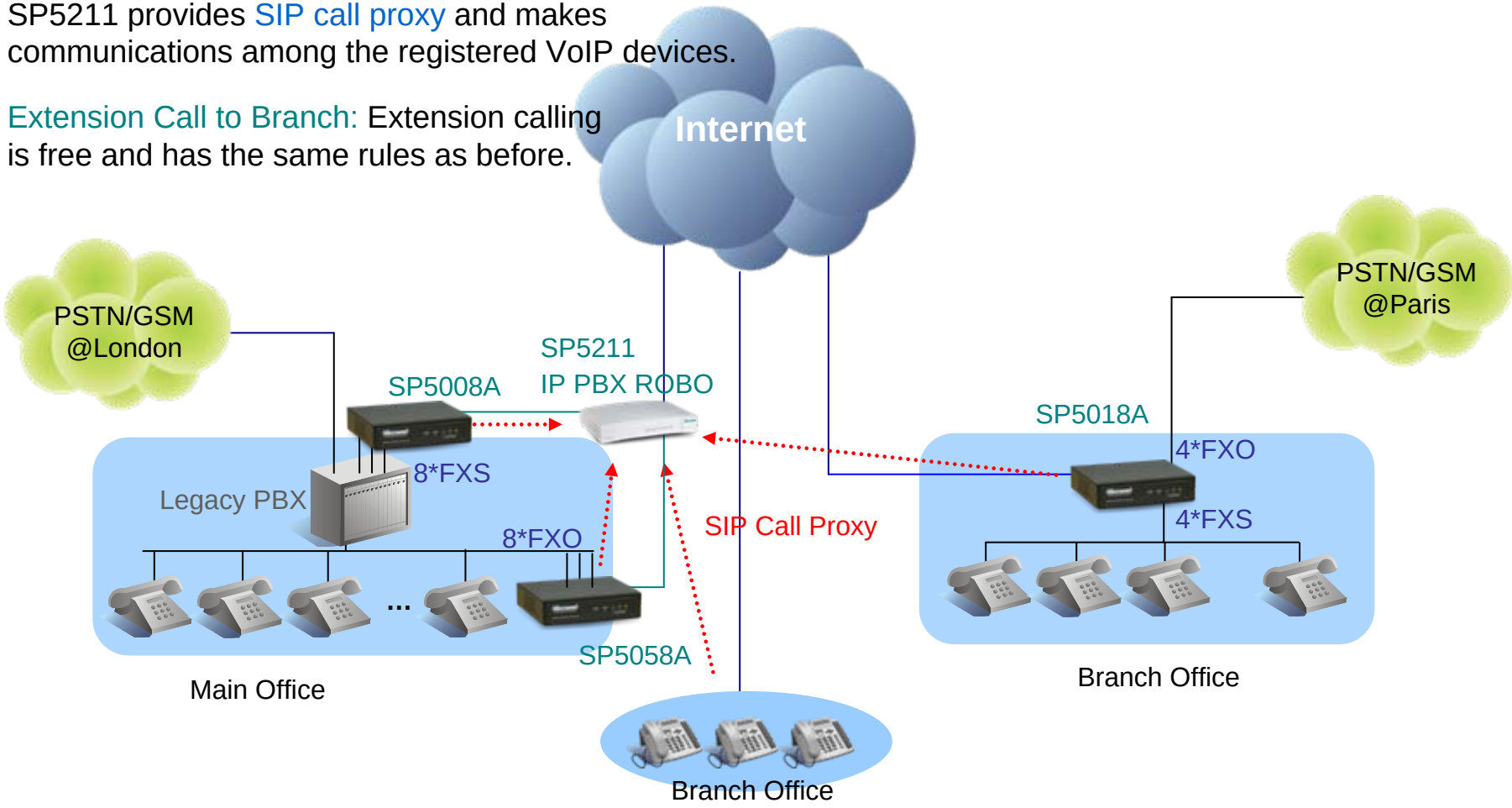
Implementation: SP5008A, SP5058A, and SP5018A are deployed on multiple sites to integrate with legacy devices. As the number of VoIP device increases, SP5211 provides SIP call proxy and makes communications among the registered VoIP devices.

Extension Call to Branch: Extension calling is free and has the same rules as before.

International Call: Phone call charge is much less and has the same rules as before.



CASE 12: VoIP Integration for Main/Branch Office, Major Upgrade

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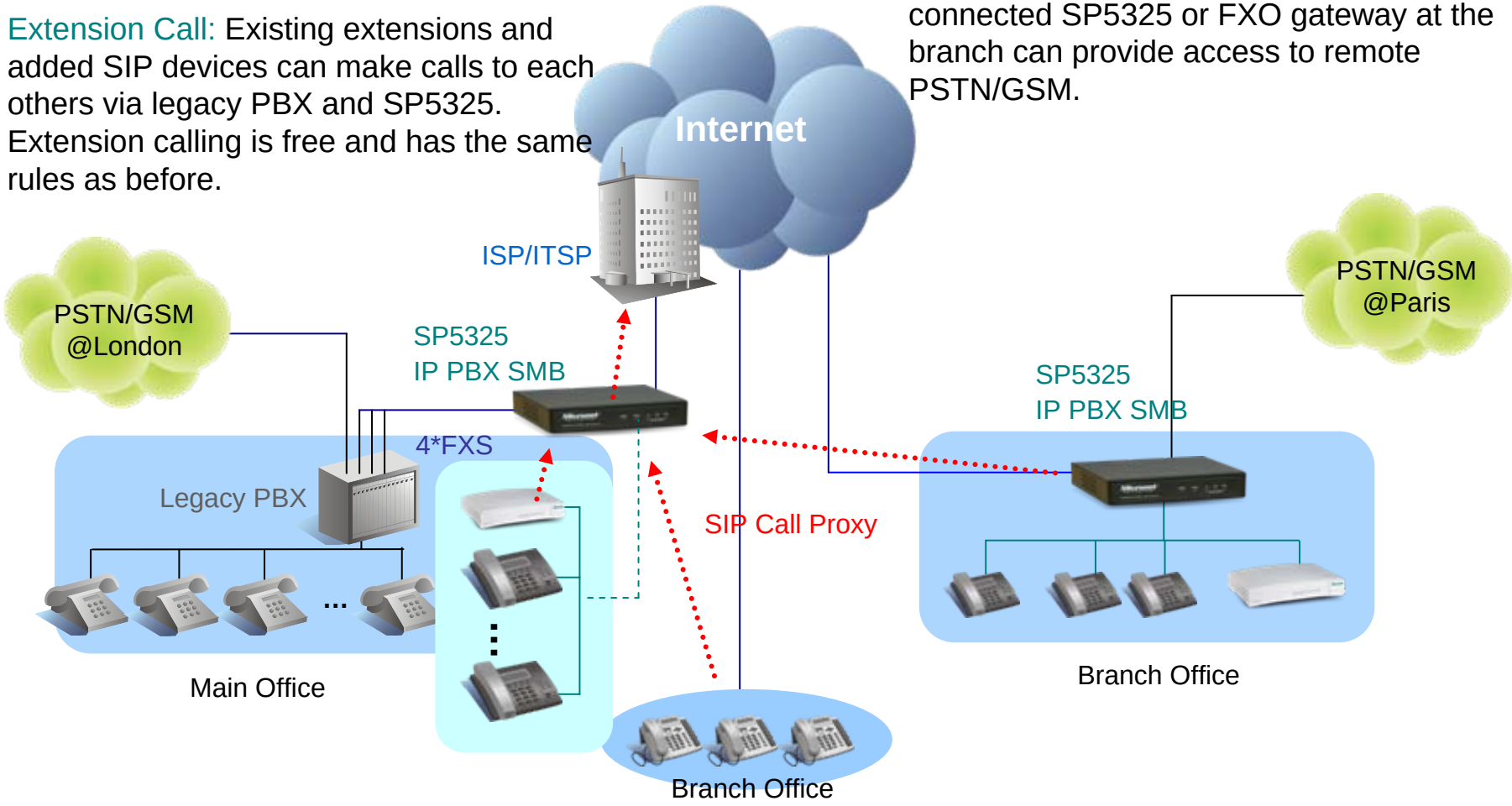
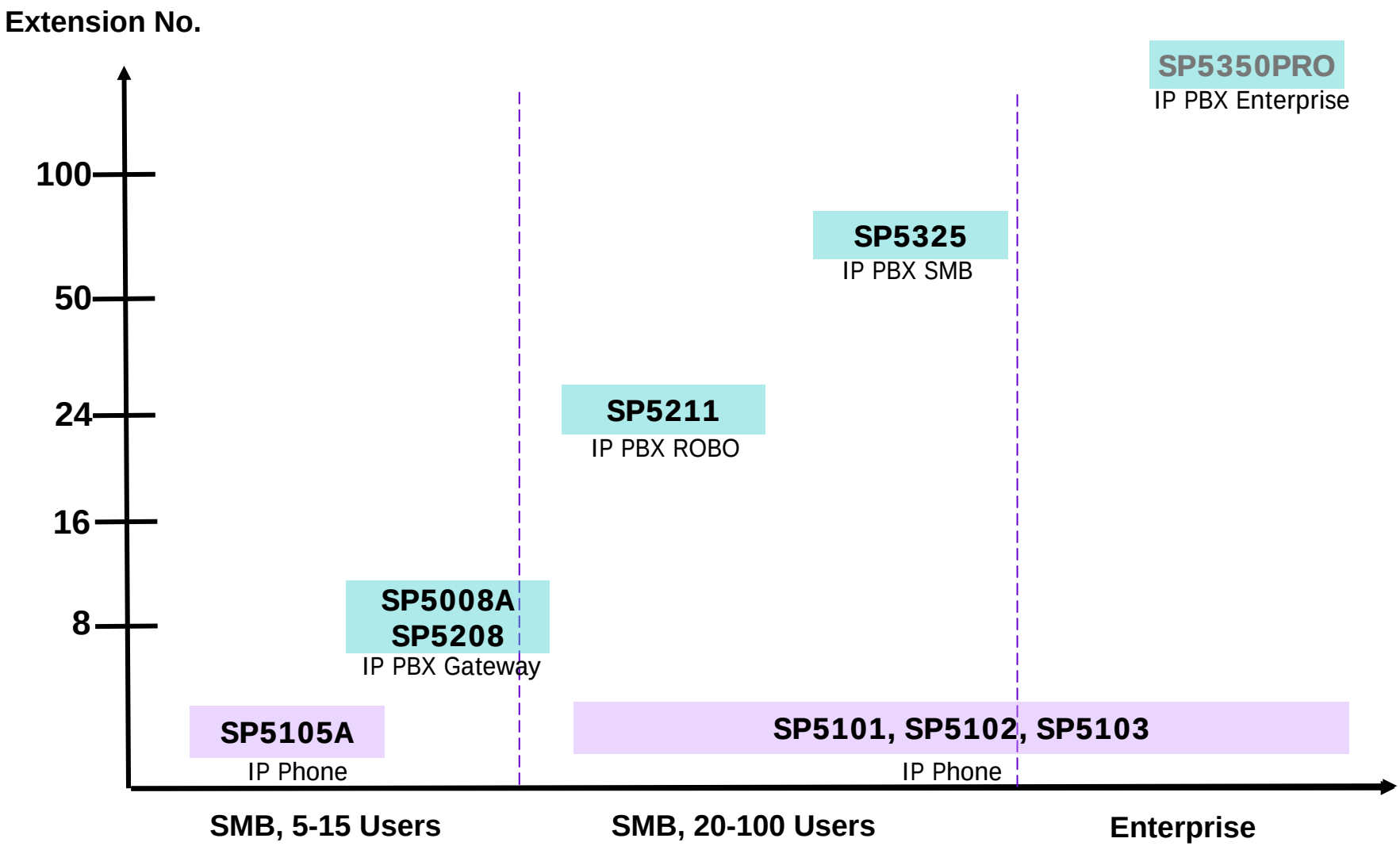
Expansion for Legacy Implementation: SP5325 connects with legacy PBX via FXS ports and SIP-enabled devices for expansion of the scale and the functionality (VoIP).

Extension Call: Existing extensions and added SIP devices can make calls to each others via legacy PBX and SP5325. Extension calling is free and has the same rules as before.

International Call: SP5325 has SIP trunk to ISP/ITSP that provides cost-saving route worldwide for outbound calling. In addition, connected SP5325 or FXO gateway at the branch can provide access to remote PSTN/GSM.



IP PBX / IP Phone



IP PBX

IP PBX Gateway

IP PBX ROBO

IP PBX SMB

Model No.	Max. Extension	Concurrent RTP Call	Analog Port	PBX Features	Protocol
New **SP5208 IP PBX Gateway for SMB**	6	6	6 FXS + 2FXO	- Call Control (Legacy PBX Functions) - Prefix Routing - Simple IVR - PSTN Backup for Power/Register Failure	SIP
SP5211 IP PBX (Asterisk)	30	13	-	- Call Control (Legacy PBX Functions) - Prefix Routing - Auto Attendant - Voicemail System	SIP
New **SP5325* IP PBX for SMB**	70	25	4 FXS 4 FXO 2 FXS + 2 FXO 1 FXS + 3 FXO	- Call Control (Legacy PBX Functions) - Prefix Routing - Auto Attendant (3-layer IVR) - Voicemail System - 8-Party Meet-Me Conference - Auto-Provisioning for IP Phone	SIP

Please refer to Product Solution Guide or User's Manual for more information. * To be available in Q2, 2009

IP PBX Gateway

Model No.: SP5208

SIP
Standards Compliant

6 FXS
Extension Calling

PBX
Smart Design

IVR
Automated Attendant

2 FXO
PSTN Backup

- IETF SIP standards compliant
- Support extension calling among 6 FXS ports
- Support 2 FXO ports to connect with PSTN lines
- Support SIP trunk for inbound/outbound SIP calls via ITSP/ISP
- Support single-account registration for representative number
- Support prefix routing for outbound calling via PSTN or SIP trunk
- Support simple IVR / automated attendant and operator setting
- Support PSTN backup for power/register failure
- Provide convenient call features: call hold, call transfer, call forward, Speed dial, emergency call (PSTN), etc

Please refer to Product Solution Guide or User's Manual for more information.

SP5208
IP PBX Gateway,
6 FXS + 2 FXO (PSTN)

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IP PBX

Model No.: SP5211, SP5211-5, SP5211-10

30
Extensions Registration

13
Concurrent RTP Calls

Asterisk
Embedded

IVR
Automated Attendant

VM
Voicemail System

- Support 30 subscribers and 13 concurrent calls
- PSTN trunking via SP5054A/S or SP5058A
- PBX features: Call hold, call transfer, call waiting, 3-way conference, call forward, music on hold/transfer, Meet-Me conference, busy call back, call park, call pickup, broadcast, DID, DOD, speed dial, BLF, CLIR, DND, features access code, call monitor, call history, CDR, etc.
- Flexible prefix call rules via the setting of call route and route group
- Auto attendant: Configurable IVR, work time setting, key to reach operator
- Voicemail system: Voicemail Notification by MWI, 1GB voicemail storage, voicemail to email, voicemail box access control (PIN Code)

Please refer to Product Solution Guide or User's Manual for more information.

SP5211
IP PBX ROBO, SIP

SP5211-5
IP PBX System, 5-Phone Kit (with SP5101)

SP5211-10
IP PBX System, 10-Phone Kit (with SP5101)

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IP PBX SMB

Model No.: SP5325, SP5325-10, SP5325-30, SP5325-50

70
Extensions Registration

25
Concurrent RTP Calls

FXS/FXO
4 POTS Interfaces

Meet-Me
Multi-Room Conference

IVR
Automated Attendant

VM
Voicemail System

IP Phone Provision

Keypad
Client-end Configuration

- Support 70 subscribers and 25 concurrent RTP calls
- Support 4 ports of FXS / FXO interface (4O, 1S3O, 2S2O, 4S)
- Rich PBX features
- Meet-Me multi-room conference: up to 8 parties (max.)
- Call route and route group for flexible prefix call rules
- 3-layer IVR with work time setting
- Voicemail system: Voicemail notification by MWI/VMWI, 256MB storage, voicemail box access control, voicemail to email
- Auto provisioning for IP Phone (SP5103 only) deployment and maintenance
- Extension configuration and registration via IP Phone keypad (SP5103 only)
- PSTN Backup: PSTN bypass if network or system fails (SP5325/2S2O and SP5325/1S3O only)

SP5325
IP PBX SMB

SP5325-10
IP PBX System, 10-Phone Kit (with SP5103)

SP5325-30
IP PBX System, 30-Phone Kit (with SP5103)

SP5325-50
IP PBX System, 50-Phone Kit (with SP5103)

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IP Phone

Dual-Mode Phone

SIP Phone

SIP Phone

Consumer SIP Phone

Model No.	LAN	WAN	PC Port	Headset	Concurrent Call	Protocol
SP5100 **Dual VoIP/PSTN Phone**	1	-	1	-	1	H.323, SIP
SP5101 **SIP Phone**	1	-	1	Optional	2	SIP
SP5102 **SIP Phone with PoE**	1	-	1	Optional	2	SIP
SP5103 **SIP Phone (for SP5325)** New	1	1	-	-	2	SIP
SP5105A* **Consumer SIP Phone**	1	1	-	-	2	SIP

Please refer to Product Solution Guide or User's Manual for more information. * To be available in Q3, 2009

PSTN/VoIP Phone

Model No.: SP5100, SP5100/S

SIP/H.323
Dual Protocols

Lifeline
PSTN Backup

Voice Quality

- Dual protocols support: SIP or H.323
- 1 PC port to connect with PC without extra wiring closet
- Provide RJ-11 port for usual Telephone Network
- Dial Plan: Phone address book, Prefix routing, Digit manipulation
- Call features: Call hold, Call transfer, Call forward, Hotline, Speed dial
- Excellent voice quality:
  - Codec: G.711A/ μ -law, G.723.1, G.729a, G.729b
  - VAD, CNG, Acoustic Echo Cancellation, Bad Frame Interpolation, Dynamic Jitter Buffer
  - ToS setting for QoS

SP5100
PSTN/VoIP Phone

Please refer to Product Solution Guide or User's Manual for more information.

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Business SIP Phone

Model No.: SP5101, SP5102

SIP Standards Compliant

2-Line Business Phone

Voice Quality

802.3af Power over Ethernet

- IETF SIP (RFC 3261) compliant
- 1 PC port to connect with PC without extra wiring closet, and 1 RJ-11 port to connect with headset
- Call features: Call hold, Call transfer, Call waiting, Call forward, 3-way conference, Hotline, Speed dial, DND
- Excellent voice quality:
  - Codec: G.711A/ μ -law, G.723.1, G.729, G.729a, G.729b
  - VAD, CNG, Acoustic Echo Cancellation, Bad Frame Interpolation, Dynamic Jitter Buffer
  - ToS setting for QoS
- Support 802.3af Power over Ethernet (SP5102 only)

SP5101
SIP Phone

SP5102
SIP Phone with PoE

Please refer to Product Solution Guide or User's Manual for more information.

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Business SIP Phone

Model No.: SP5103

SIP
Standards Compliant

2-Line
Business Phone

Voice Quality

IP PBX
Provision

- IETF SIP (RFC 3261) compliant
- Multi-line business phone
- Support up to 3 SIP accounts
- Rich call features: call hold, call transfer, 3-way conference, call waiting, call forward, online call recording, call pickup, call park, DND, etc
- Call details list and MWI (message waiting indicator)
- Excellent voice quality:
  - Codec: G.711A/ μ -law, G.723.1, G.729, G.729a, G.729b
  - VAD, CNG, Acoustic Echo Cancellation, Bad Frame Interpolation, Dynamic Jitter Buffer
  - ToS setting for QoS
- Auto Provisioning on boot (co-working with SP5325 IP PBX)

Please refer to Product Solution Guide or User's Manual for more information.

SP5103
SIP Phone

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Consumer SIP Phone

Model No.: SP5105A

SIP
Standards Compliant

2-Line
Call Concurrently

Voice Quality

- IETF SIP (RFC 3261) compliant
- Call features: Call hold, Call transfer, Call waiting, Call forward, 3-way conference, Hotline, Speed dial, DND
- Dial Plan: Phone address book, Prefix routing, Digit manipulation
- Excellent voice quality:
  - Codec: G.711A/ μ -law, G.723.1, G.726, G.729a, G.729b
  - VAD, CNG, Acoustic Echo Cancellation, Bad Frame Interpolation, Dynamic Jitter Buffer
  - DSCP setting for QoS

Please refer to Product Solution Guide or User's Manual for more information.

SP5105A
Consumer SIP Phone

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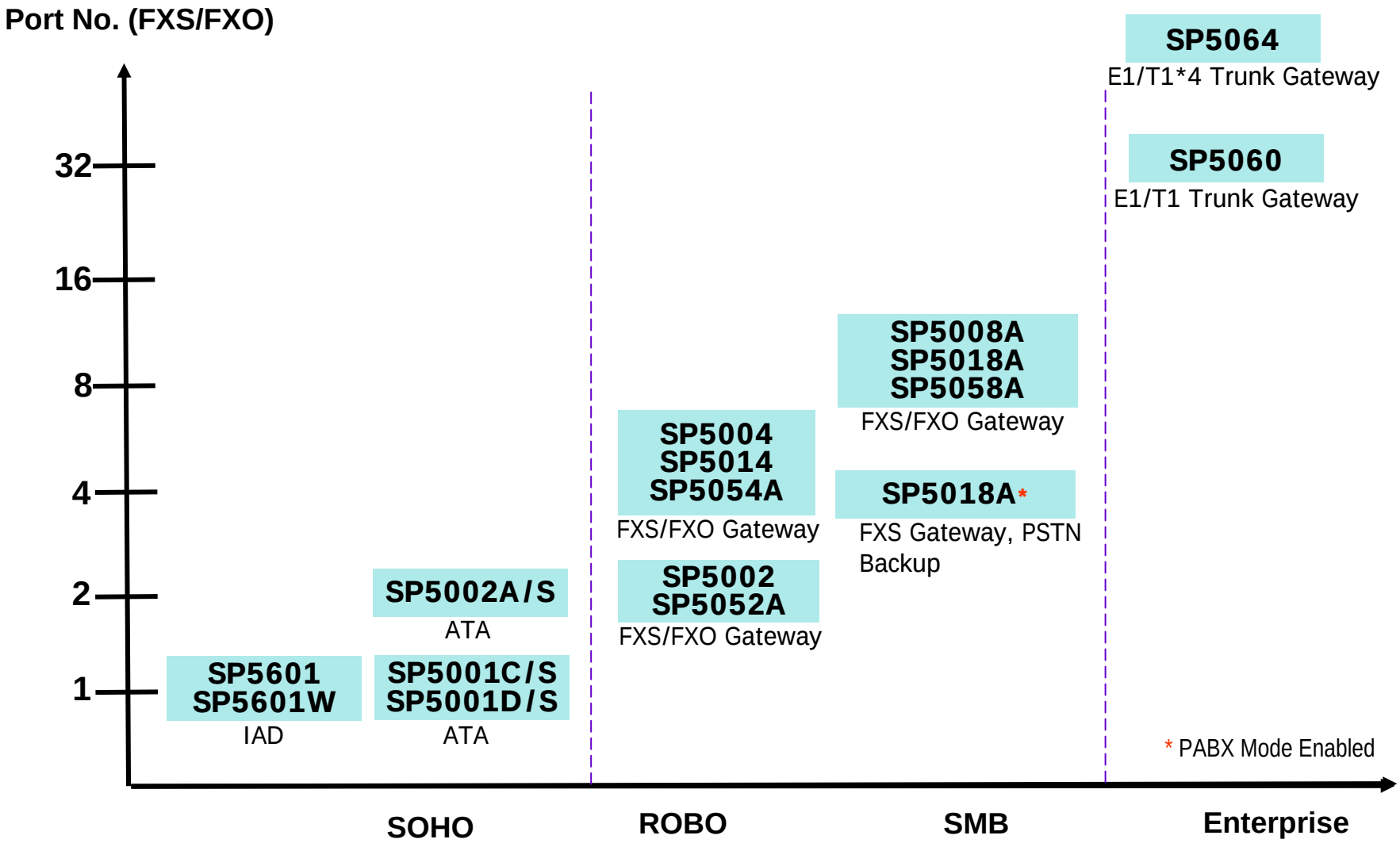
VoIP Gateway



Analog Gateway, FXS

SIP FXS Gateway (ATA)

2/4-Port H.323/SIP FXS Gateway

8-Port VoIP FXS Gateway

	Model No.	FXS	PSTN Lifeline	NAT	WAN/LAN	Protocol
SIP FXS Gateway (ATA)	**SP5001C/S**	1	-	Yes	1/1	SIP
	SP5001D/S	1	1	Yes	1/1	SIP
	SP5002A/S	2	-	Yes	1/1	SIP
VoIP FXS Gateway	**SP5002**	2	-	-	0/2	SIP, H.323
	SP5004	4	-	-	0/2	SIP, H.323
	SP5008A *New*	8	-	Yes	1/4	SIP

SIP FXS Gateway (ATA)

Model No.: SP5001C/S, SP5001D/S, SP5002A/S

SIP
Standard Compliant

Dial Plan
Flexible Dial Rules

QoS
Voice Quality

- IETF SIP (RFC 3261) compliant
- 1/2 FXS ports to connect with standard telephone or fax machine, and 1 PSTN lifeline for backup
- Dial Plan: Phone address book, Prefix routing, Digit manipulation
- Call features: Call hold, Call transfer, 3-way conference, Call waiting, Call forward, Hotline, Speed dial, DND
- Ensure voice quality:
  - Codec: G.711A/ μ -law, G.729a/b, G.726
  - VAD, CNG, Echo Cancellation, Dynamic Jitter Buffer
  - DSCP/CoS setting for QoS

SP5002A/S
VoIP FXS Gateway,
2 FXS, SIP

Please refer to Product Solution Guide or User's Manual for more information.

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VoIP FXS Gateway

Model No.: SP5002, SP5002/S, SP5004, SP5004/S

SIP/H.323
Dual Protocols

Dial Plan
Flexible Dial Rules

Voice Quality

- Dual protocols support: SIP or H.323
- 2/4 FXS ports to connect with standard telephone, fax machine, or PBX's trunk
- Dial Plan: Phone address book, Prefix routing, Digit manipulation
- Call features: Call hold, Call transfer, Call forward, Hotline, Speed dial
- Excellent voice quality:
  - Codec: G.711A/ μ -law, G.723.1, G.729a, G.729b, G.729ab
  - VAD, CNG, Acoustic Echo Cancellation, Bad Frame Interpolation, Dynamic Jitter Buffer
  - DSCP setting for QoS

Please refer to Product Solution Guide or User's Manual for more information.

SP5004/S
VoIP FXS Gateway,
4 FXS, SIP

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VoIP FXS Gateway

Model No.: SP5008A, SP5018A*

SIP
Standards Compliant

4/8 FXS
Extension Calling

Dial Plan
Flexible Dial Rules

PBX
Smart Design

Lifeline
PSTN Backup

- IETF SIP standards compliant
- Supports 4/8 FXS ports for connecting with legacy PBX, phone set, or fax machine
- Support extension calling among 4/8 FXS ports
- Support single-account registration on FXS ports for representative number
- Support SIP trunk for inbound/outbound SIP calls via ITSP/ISP
- Provide convenient call features: call hold, call transfer, call forward, hotline, warm line, speed dial, anonymous call, DID, DOD, DND, P2P call, etc
- Excellent voice quality
- Support PABX mode by enabling 4 PSTN lifelines for legacy PABX function upgrade (SP5018A only)

SP5008A
VoIP FXS Gateway,
8 FXS

SP5018A*
VoIP FXS Gateway,
4 FXS + 4 PSTN

* PABX Mode Enabled

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Analog Gateway, FXS

ADSL2+ VoIP Router (IAD)

Wireless ADSL2+ VoIP Router (IAD)

	Model No.	FXS	FXO	Wireless	ADSL2+	LAN ports	Protocol
ADSL2+ VoIP Router (IAD)	**SP5601**	1	PSTN Lifeline	-	Yes	4	SIP
	SP5601W	1	PSTN Lifeline	Yes	Yes	4	SIP

ADSL2+ VoIP Router (IAD)

Model No.: SP5601, SP5601W

SIP
Standards Compliant

ADSL2+
Modem Embedded

WiFi
Cordless Access

Firewall
Secured Home Network

TR-069
CPE WAN Management

- IETF SIP (RFC 3261) compliant
- 1 FXS port to connect with standard telephone and 1 PSTN lifeline for backup
- Support ADSL2+ standards and provide up to 24M/2M Internet access rate
- Full-function broadband router with 4 LAN ports
- Support 802.11b/g standards and Provide up to 54M wireless operation rate (SP5601W only)
- Call features: Call hold, Call transfer, Call waiting, Speed dial, Dial Plan, DND
- Rich Firewall security features: Packet Inspection, DoS attack prevention, and URL Content Filtering
- Allow ISP remotely manage SP5601/SP5601W via TR-069 Auto-Provisioning functions

Please refer to Product Solution Guide or User's Manual for more information.

SP5601W
Wireless ADSL2+ VoIP Router, SIP, 1 FXS, 1PSTN, 4 LAN

SP5601
ADSL2+ VoIP Router, SIP, 1 FXS, 1PSTN, 4 LAN

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Analog Gateway, FXS+FXO / FXO

4-Port H.323/SIP FXS/FXO Gateway

8-Port VoIP FXS/FXO Gateway

	Model No.	FXS	FXO	PSTN Lifeline	NAT	WAN/LAN	Protocol
FXS/FXO Gateway	**SP5014**	2	2	Yes (for 2 FXO)	-	0/2	SIP, H.323
	SP5018A *New*	4	4	Yes (for 4 FXO)	Yes	1/4	SIP
FXO Gateway	**SP5052A**	-	2	-	-	0/2	SIP, H.323
	SP5054A	-	4	-	-	0/2	SIP, H.323
	SP5058A *New*	-	8	-	Yes	1/4	SIP

VoIP FXS/FXO Gateway

Model No.: SP5014, SP5014/S

SIP/H.323
Dual Protocols

Lifeline
PSTN Backup

Dial Plan
Flexible Dial Rules

Voice Quality

- Dual protocols support: SIP or H.323
- 2 FXS ports to connect with standard telephone, fax machine, or PBX's trunk
- 2 FXO ports to connect PSTN or PBX's extension, and support PSTN bypass ports for backup
- FXO ports support detection of polarity reversal and disconnect tone
- Dial Plan: Phone address book, Prefix routing, Digit manipulation
- Call features: Call hold, Call transfer, Call forward, Hotline, Speed dial, FXO Line hunting method
- Excellent voice quality:
  - Codec: G.711A/ μ -law, G.723.1, G.729a/b
  - VAD, CNG, Acoustic Echo Cancellation, Bad Frame Interpolation, Dynamic Jitter Buffer
  - DSCP setting for QoS

Please refer to Product Solution Guide or User's Manual for more information.

SP5014/S
VoIP FXS/FXO Gateway,
2 FXS + 2 FXO, SIP

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VoIP FXS/FXO Gateway

Model No.: SP5018A

SIP
Standards Compliant

4 FXS / 4 FXO

Dial Plan
Flexible Dial Rules

Voice Quality

Lifeline
PSTN Backup

- IETF SIP standards compliant
- Supports 4 FXS ports for connecting with legacy PBX, phone set, or fax machine
- Support extension calling among 4 FXS ports
- Support SIP trunk for inbound/outbound SIP calls via ITSP/ISP
- Support rich call features: call hold, call transfer, call forward, hotline, warm line, speed dial, anonymous call, P2P call, etc
- Support PSTN prefix routing
- Excellent voice quality
- Support PABX mode by enabling 4 PSTN lifelines for legacy PABX function upgrade
- Support detection of disconnect tone, polarity reversal, and loop current drop (zero voltage) on FXO ports

SP5018A
VoIP FXS/FXO Gateway,
4 FXS + 4 FXO

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VoIP FXO Gateway

Model No.: SP5052A, SP5052A/S, SP5054A, SP5054A/S

SIP/H.323
Dual Protocols

Dial Plan
Flexible Dial Rules

Voice Quality

- Dual protocols support: SIP or H.323
- 2/4 FXO ports to connect with PSTN or PBX's extension
- Dial Plan: Phone address book, Prefix routing, Digit manipulation
- Call features: Hotline, FXO Line hunting method
- Support detection of power denial, polarity reversal and disconnect tone
- Excellent voice quality:
  - Codec: G.711A/ μ -law, G.723.1, G.729a/b
  - VAD, CNG, Acoustic Echo Cancellation, Bad Frame Interpolation, Dynamic Jitter Buffer
  - DSCP setting for QoS

SP5054A/S
VoIP FXO Gateway,
4 FXO, SIP

Please refer to Product Solution Guide or User's Manual for more information.

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VoIP FXO Gateway

Model No.: SP5058A

SIP/H.323
Dual Protocols

8 FXO
High-Port Density

Voice Quality

- IETF SIP standards compliant
- Support POTS interfaces: 8 FXO ports
- Support rich call features: call hold, call transfer, call forward, P2P call, etc
- Excellent voice quality:
- Support detection of disconnect tone, polarity reversal, and loop current drop (zero voltage) on FXO ports

SP5058A
VoIP FXO gateway, 8 FXO

Please refer to Product Solution Guide or User's Manual for more information.

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Digital Gateway

VoIP E1/T1 Trunk Gateway

Model No.	E1/T1 Trunk	Call Control	In-Trunk Hunting	IVR	Billing System	Protocol
SP5060	1	Yes	Yes	Yes	RADIUS Embedded AAA	SIP, H.323
SP5064	4	Yes	Yes	Yes	RADIUS Embedded AAA	SIP, H.323

E1/T1 Trunk Gateway

Model No.: SP5060, SP5064

SIP/H.323
Dual Protocols

Dial Plan
Flexible Dial Rules

IVR
Automated Attendant

AAA
Fully Support

- Dual protocols support: SIP and H.323
- 1 or 4 E1/T1 trunk ports to connect PSTN or PBX
- Carrier-grade device
- Support PSTN signaling: E1 ISDN PRI, E1 SS7 ISUP, E1 CAS DTMF/ R2, T1 ISDN PRI, T1 CAS
- Intelligent PSTN call routing and in-trunk hunting
- Flexible digit manipulation plan
- Provide built-in IVR and call flow controller
- Support RADIUS authorization and billing message
- Suitable for pre-paid or post-paid services

SP5060/SP5064 VoIP E1/T1 Trunk Gateway, 1/4 trunk ports

Please refer to Product Solution Guide or User's Manual for more information.

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SIP Proxy Server

SIP Proxy Server

Model No.	Subscriber	Concurrent Call	Concurrent NAT RTP Call	SIP Register	Proxy	AAA	Protocol
SP5210	2000	200	100	Yes	Yes	RADIUS	SIP
SP5210-200	200	50	50	Yes	Yes	RADIUS	SIP
SP5210-500	500	100	100	Yes	Yes	RADIUS	SIP
SP5210K	20000	2000	384	Yes	Yes	RADIUS	SIP

SIP Proxy Server

Model No.: SP5210 Series

SIP
Standards Compliant

Prefix Routing

NAT
Traversal

AAA
Fully Support

- IETF SIP (RFC 3261) compliant
- Carrier-grade device
- Support up to 20,000 subscribers and 2,000 concurrent calls
- Support intelligent call routing and NAT pass-thru
- Provide rich advanced call features
- Support Authorization, Authentication, and Accounting (AAA) mechanism

SP5210 Series
SIP Proxy Server

Please refer to Product Solution Guide or User's Manual for more information.

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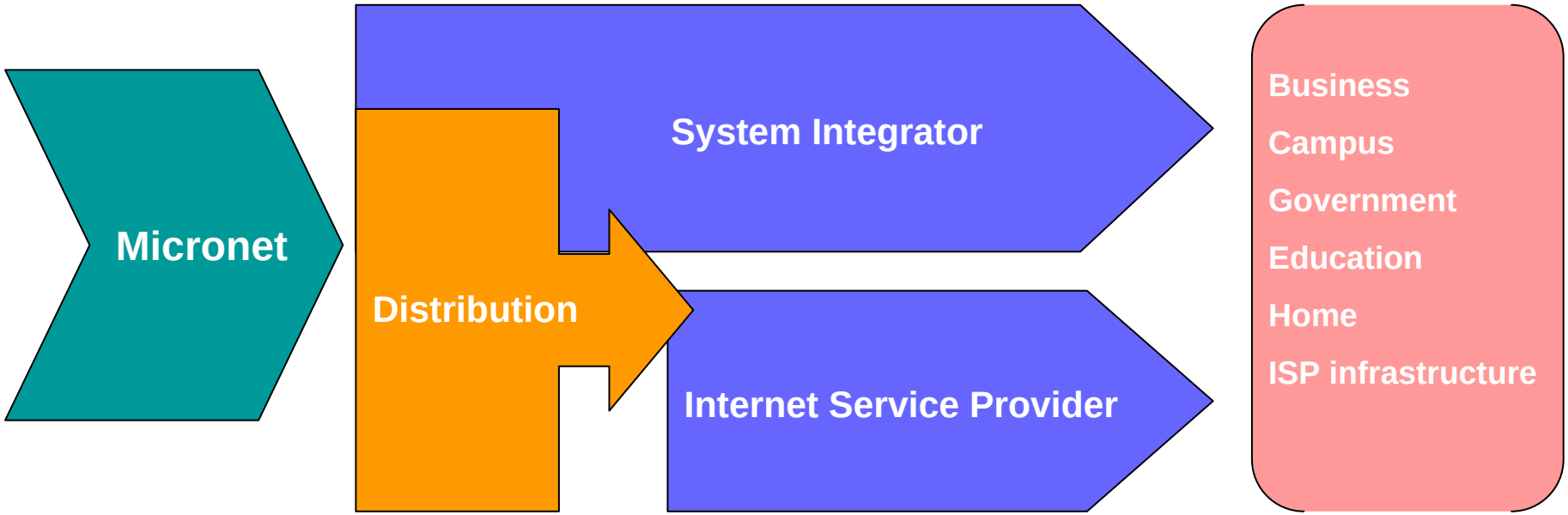
Selling Channels



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